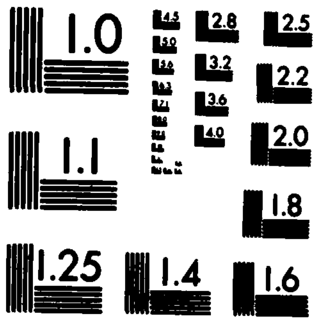
AD-A125 360 COMPUTER RECOGNITION OF PHONETS IN SPEECH(U) AIR FORCE INST OF TECH WRIGHT-PATTERSON AFB OH SCHOOL OF ENGINEERING D L MARTIN DEC 82 AFIT/GE/EE/82D-46 1/3

UNCLASSIFIED F/G 9/2 NL

M-2



MICROCOPY RESOLUTION TEST CHART
NATIONAL BUREAU OF STANDARDS-1963-A

AD A125360 ①

Computer Recognition of Phonets in Speech

Thesis

AFIT/GE/EE/82D-4 Dan Martin
Captain, USAF

DEPARTMENT OF THE AIR FORCE
AIR UNIVERSITY (ATC)

AIR FORCE INSTITUTE OF TECHNOLOGY

Wright-Patterson Air Force Base, Ohio

REPORT DOCUMENTATION PAGE

READ INSTRUCTIONS BEFORE COMPLETING FORM

1. REPORT NUMBER AFIT/GE/EE/82D-46	2. GOVT ACCESSION NO. AD-A125376	3. RECIPIENT'S CATALOG NUMBER
4. TITLE (and Subtitle) Computer Recognition of Phonets In Speech		5. TYPE OF REPORT & PERIOD COVERED MS Thesis
		6. PERFORMING ORG. REPORT NUMBER
7. AUTHOR(s) Daniel L. Martin, Captain, USAF		8. CONTRACT OR GRANT NUMBER(s)
9. PERFORMING ORGANIZATION NAME AND ADDRESS Air Force Institute of Technology (AFIT/EN) Wright-Patteron AFB OH 45433		10. PROGRAM ELEMENT, PROJECT, TASK AREA & WORK UNIT NUMBERS
11. CONTROLLING OFFICE NAME AND ADDRESS		12. REPORT DATE December 1982
		13. NUMBER OF PAGES 226
14. MONITORING AGENCY NAME & ADDRESS(if different from Controlling Office)		15. SECURITY CLASS. (of this report) Unclassified
		15a. DECLASSIFICATION/DOWNGRADING SCHEDULE

16. DISTRIBUTION STATEMENT (of this Report)

Approved for public release; distribution unlimited.

17. DISTRIBUTION STATEMENT (of the abstract entered in Block 20, if different from Report)

18. SUPPLEMENTARY NOTES

Approved for public release: IAW AFR 190-17.
[illegible]
LYNN E. WOLAVER
Dean for Research and Professional Development
Air Force Institute of Technology (ATC)
Wright-Patterson AFB OH 45433

4 JAN 1983

19. KEY WORDS (Continue on reverse side if necessary and identify by block number)
- phoneme recognition by prototype matching
- speech synthesis by phoneme sounds
- phoneme recognition by average magnitude differencing
- continuous speech recognition
(continued on reverse side)

20. ABSTRACT (Continue on reverse side if necessary and identify by block number)
This project generated phonetic units, termed "phonets," from digitized speech files. The time file was converted to feature space using the Fourier Transform, and phonet occurrences were detected using Minkowski One and Two distance measures. Phonet matches were detected and ranked for each phonet compared against a template file. Phonet short-time energy was included in the output files. An algorithm was developed to partition feature space and its performance was evaluated.

19. (Cont'd)
- phoneme recognition by spectrogram matching
- analysis of phoneme sounds in continuous speech

①

GE/EE/82D-46

Computer Recognition of Phonets in Speech

Thesis

AFIT/GE/EE/82D-46 Dan Martin
Captain, USAF

GE/EE/82D-46

Computer Recognition of Phonets In Speech

Thesis

Presented to the Faculty of the School of Engineering

of the Air Force Institute of Technology

Air University

In Partial Fulfillment of the

Requirements for the Degree of

Master of Science

By

Daniel L. Martin, B.S.E.E., B.S.M.A.

Captain USAF

Graduate Electrical Engineering

December 1982

ACKNOWLEDGEMENTS

I wish to express my appreciation and gratitude to Dr. Mathew Kabrisky, my thesis advisor, for his help and encouragement through the course of this effort. I also want to thank Major Larry Kizer and Dr. Peter Maybeck, members of my thesis committee, for their assistance and constructive criticism.

PREFACE

The purpose of this project was to generate and detect features in connected speech. It is a part of the larger speech recognition problem.

Phonetic units, which we called observations and phonets, were generated from connected speech. The time file was converted to feature space using the Fourier Transform. Phonet occurrences were detected using Minkowski One and Two distance measures. Phonet matches were ranked in order from minimum to maximum distance between each input phonet and each phonet in the template file.

An algorithm was developed to partition feature space into classes of phonetic units. The central tendency of each class was considered to be a template against which observations were to be compared. The algorithm is adaptive in that a supervisory capability is provided to decide, on the basis of detection results and class variability, if class descriptors should be modified. Algorithm performance in additive, white, guassian noise was evaluated and decision parameters were related to probability of classification error.

TABLE OF CONTENTS

LIST OF FIGURES

LIST OF TABLES

ABSTRACT

"Phonet Detection in Connected Speech"

This project generated phonetic units, termed "phonets," from digitized speech files. The time file was converted to feature space using the Fourier Transform, and phonet occurrences were detected using Minkowski One and Two distance measures. Phonet matches were detected and ranked for each phonet compared against a template file. Phonet short-time energy was included in the output files. An algorithm was developed to partition feature space and its performance was evaluated.

I. INTRODUCTION.

BACKGROUND:

This project concerns the generation and detection of features in connected speech. It is a part of a larger effort directed towards connected speech recognition.

We have divided the speech recognition problem into six parts:

(1) Digitization of the acoustic signal,

(2) Feature extraction,

(3) Partitioning of the feature space into classes,

(4) Feature detection,

(5) Word recognition from detected features, and

(6) Concept recognition from recognized words.

Falkey (Ref 2) and Seelandt (Ref 1) have worked on Parts I, II, and IV of the problem. This project extends Seelandt's work on feature detection and offers an algorithm for partitioning feature space into classes. Montgomery (Ref 10) has worked on Part V of this problem and Part VI has yet to be addressed.

In the remainder of this chapter, we mention works in the literature which influenced the development of this project and state its scope.

Felkey was able to generate spectrograms from digitized speech (Ref 2). This capability was extended by Seelandt (Ref 1) into a Speech Sound Analysis Machine (SSAM) which gave an operator the ability to listen to speech segments chosen from a spectrogram displayed on a Tektronix 4010-1 Graphics Terminal. Seelandt was able to use his machine to choose phonetic units which he thought might be detectable in connected speech. He was able to construct template files of these units and then to detect similar units in connected speech. Detection was accomplished by calculating a Minkowski metric of order one (Ref 1 and 9) between each phonetic unit and spectrum of consecutive time slices from the input speech. The template which matched the speech segment in the sense of minimum distance was deemed the detected phoneme.

Felkey and Seelandt both applied the Fourier Transform and extracted features in the spectral domain. Other transformations from measurement space can be applied. DeSouza and Thomson derived likelihood ratio statistics for the case in which two estimated Linear Predictive Coding (LPC) vectors are being compared (Ref 3). They also investigated the distribution and sensitivity of other LPC distance measures.

Kassam has compiled an alphabetical listing of papers from the engineering literature on nonparametric detection theory and application (Ref 4). Nonparametric detectors

12

are based on statistical hypothesis testing principles for situations where parametric statistical models cannot be specified for the observation under the null hypothesis. The usual performance characteristic is minimum Type I error probability or false-alarm rate. Kassam also states that robust procedures, where not only the false-alarm probability but also the power of the test is considered in defining a performance criterion, may be applicable to situations where parametric models cannot be assumed.

Lainiotis (Ref 5) has studied adaptive systems for detection and pattern recognition as well as for feature extraction. He states that for the supervised learning case, optimal adaptive systems are realizable in a partitioned form which consists of a linear, nonadaptive part made up of a bank of Kalman filters, and a nonlinear, adaptive part made up of probability computers. He offers relatively tight upper and lower bounds to the probability of error in the feature extraction problem in terms of functionals such as the Bhattacharya coefficient.

Work done at Johns Hopkins University points out that the Fast Fourier Transform (FFT) is an extremely robust procedure for large time-band width signals (Ref 7). The report comments that the FFT is robust enough to give good performance on an error probability basis on random input time series with probability distributions so extreme as to be unlikely to occur in nature. The price for this robustness is reduced power; that is, the probability of making a correct binary decision conditioned on knowledge of the signal sent, according to the report.

We decided to continue feature extraction in the spectral domain for the following reasons:

(1) We could build directly on the work already accomplished by Seelandt and Felkey.

(2) We could apply the already available Eclipse AP/130 Array Processor to the task of transforming the speech time series to the spectral domain.

(3) While no one transformation into feature space that we found documented in the literature was clearly a best choice for our purpose, we deemed it wise to sacrifice power for the robustness of the FFT.

SCOPE:

The goals of this project are:

(1) To generate phonetic units, which we term phonets or observations, from speech files placed on disk.

(2) To detect occurrence of members of one set of phonetic units, referred to as phonets, in a contiguous file of observations generated from speech.

(3) To apply the Eclipse AP/130 Array Processor to tasks one and two for speed.

(4) To build a sufficient degree of flexibility into the phonet generator and distance computer to allow

factors, such as FFT window point size and type, filtering, and distance rule, to be varied and their effect investigated.

(5) To identify a means for generating an adequate phonet set for our purposes.

Tasks one through four are accomplished by development of the Acoustic Analyzer: an interactive software package which accomplishes those tasks. Task five is accomplished by formulation of an algorithm for partitioning feature space into template classes. Additionally, some characteristics of phonets and their interaction with observations are illustrated.

This work was not concerned with acquiring speech or generating time files containing speech. It takes the techniques Seelandt (Ref 1:9-12) used as given and relies on speech files resident on disk.

15

II. TECHNIQUES.

TIME DOMAIN PROCESSING:

In this chapter, we discuss techniques used to implement the Acoustic Analyzer. In this section, we discuss time domain processing of the digitized speech. In the following sections, we describe the generation of what we call observations and phonets: spectral domain representations of the fundamental phonetic units, and the distance computation between observations and phonets.

Speech is digitized at an 8KHz sampling rate and stored on disk in the manner described by Seelandt (Ref 1: 9-12) before being input to the Acoustic Analyzer. Given this input file, the Acoustic Analyzer converts speech to observations and then compares them to a template file of phonets. Time domain processing is accomplished in the segment of the Acoustic Analyzer which converts speech to observations and consists of the application of a window to the time file. The operator has control of the following window parameters, which are explained below:

(1) Window type.

(2) Window size.

(3) Overlap of the 128 point window.

Two types of windows are available: a Hamming and a rectangular window. The Hamming window option was included because: (1) Seelandt used it, and (2) spectral distortion caused by the rectangular window is an undesired complication. It was desirable to include the capability to select a number of different window sizes to provide flexibility in future investigations. It is clear that a choice of window size involves a trade-off between spectral resolution and time resolution (Ref 8:260). It appears that several window sizes will need to be tried and one best chosen on the basis of recognition results. Seelandt used a 64 point window at a sample rate of 8KHz for time slices 8MSEC long (Ref 1:120). Rabiner and Schafer discuss the window effects with regard to using time slice energy as an indicator of voiced or unvoiced speech (Ref 8:122). They recommend using a window duration of from 10-20MSEC. At the 8KHz sampling rate used by Seelandt to digitize the speech files (Ref 1:10), a 10-20MSEC window is from 80-160 points wide. Also, Rabiner and Schafer show that stops and other short-lived temporal features of speech last roughly 10MSEC (Ref 8:38-60). For these two reasons, we consider the 128 point window as our primary choice, or point-of-departure for further investigations into the question of optimum window size. Because of concern that a Hamming window shape coupled with 16MSEC window duration might miss some temporal detail, we included the capability to cause the window to increment along the speech file in 64 point increments overlapping by 64 points. This option is not available for the other window sizes and does not significantly add to processing time.

OBSERVATION GENERATION:

An observation is, basically, the magnitude of the Fast Fourier Transform (FFT) of a segment of an input speech file. Spectral shaping, described later, is applied to the spectral components. It is this shaped spectrum that is compared to the phonets in the template file: a phonet being an observation assigned to the template file.

Short-time energy of the speech signal is a parameter that reflects amplitude variations between unvoiced and voiced speech segments (Ref 8:120). Rabiner and Schafer define the short-time energy as:

$$E_n = \sum_{m=-\infty}^{\infty} \left[x(m)\ w\ (n-m)\right]^2 \tag{1}$$

with:

E_n = short-time energy.

$\{x(m)\}$ = speech time sequence.

$\{w(n-m)\}$ = window time sequence of finite length.

They define the time dependent Fourier Transform as (Ref 8:251):

$$X_n(e^{jw}) = \sum_{m=-\infty}^{\infty} w(n-m)\ x(m) e^{-jwm} \tag{2}$$

with:

$\left\{X_n(e^{jw})\right\}$ = time dependent Fourier Transform.

Our Acoustic Analyzer computes a value which we call the observation energy, defined as:

$$E_n^1 = \sum_{m=-\infty}^{\infty} \left| X_m\ (e^{jw})\ p(m) \right|^2 \tag{3}$$

where:

E_n^1 = observation energy.

$\left\{p(m)\right\}$ = spectral emphasis sequence.

The emphasis sequence, $\left\{p(m)\right\}$, is equivalent to a linear, shift invariant filter with a spectral response shown in Figure 1.

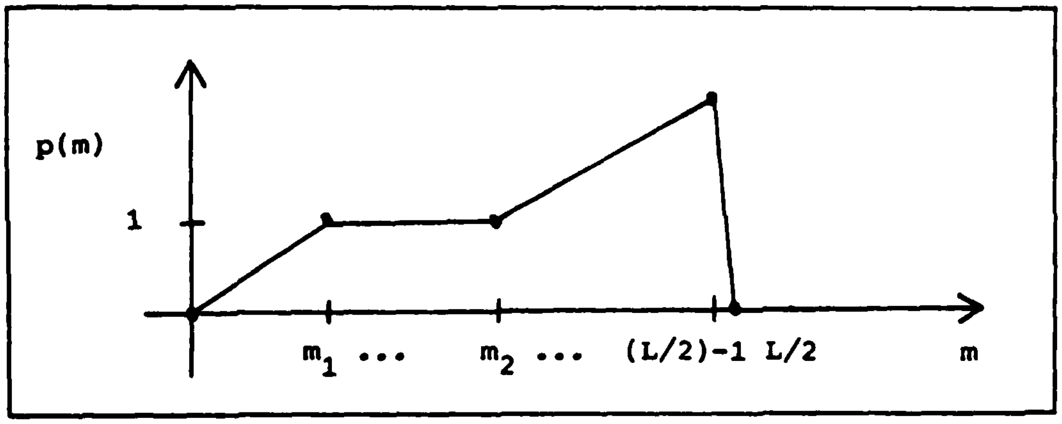


FIGURE 1. Spectrum Emphasis Filter

For an L- point window, p(m) = 0 for m = 0 and for m = L/2. Low frequency components are de-emphasized at a selectable rate up to a selectable corner frequency, m_1. High frequency components are emphasized at a selectable rate and corner frequency, m_2. Emphasis rates are selectable in units of dB/Octave. Note that the zero and L/2 points are zero. Assuming neglible energy in the speech signal at 4KHz, and assuming a DC block, the observations computed by our Acoustic Analyzer are the output of the linear system as shown in Figure 2.

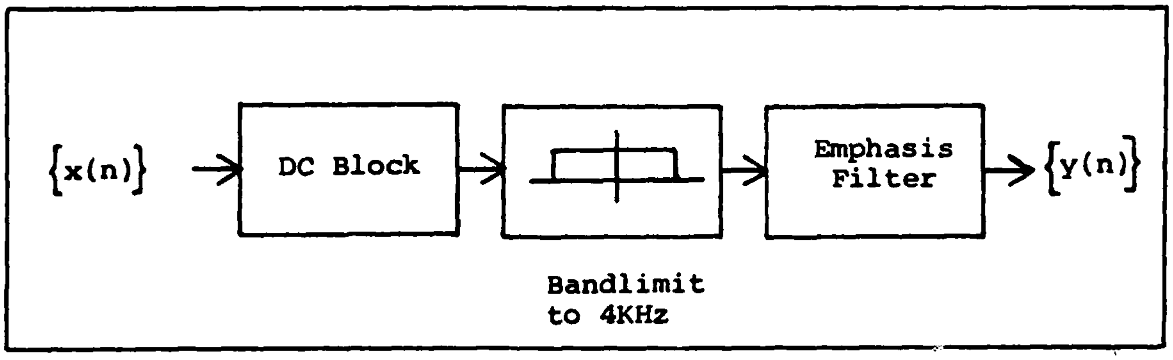


FIGURE 2. Linear System Representation of Spectral Computer In Acoustic Analyzer. $\{X(n)\}$ is the speech input.

By Parseval's theorem (Ref 9:36), the energy in the output sequence, E_n^1, is related to the energy in the input speech, E_n, under assumptions that there is: (1) no DC value to the speech input, and (2) no speech energy at or above 4KHz. Hence, the observation energy, E_n^1, computed by the Acoustic Analyzer, can be used as an indication of voiced/unvoiced speech in the same manner as the short-time energy. This value can be used by a word recognition machine to aid in that process. The output file at this point contains, for each observation, the observation energy as the first component and spectral components one through L/2 -1 in the second through L/2 component positions.

For the same reason Seelandt did (Ref 1:121), we normalized the energy in each observation to a constant value. This provided a measure of automatic gain control, in that if the only difference between two speech segments was their energy content, the components in observations produced from them would be of approximately the same amplitude. Thus, variability caused by different speech amplitudes would be reduced. To normalize the observation energy, we scaled each spectral component by the factor $10^4/\sqrt{E_n^1}$.

We observed the effectiveness of obtaining automatic gain control through energy normalizing by including the capability to scale the spectral components by the factor $10^4/E_n^1$, rather than energy normalizing. We generated two time files containing tones at 200Hz, 1KHz, and 2.5KHz, each at an amplitude of 50 units in File A and 200 units in File B. Figure 3 is an observation from File A and Figure 4 is from File B; both energy normalized. In these and the next two figures, the value plotted in position zero along the horizontal axis is the energy value, $\sqrt{E_n^1}$. One can see that the amplitudes of the spectral components

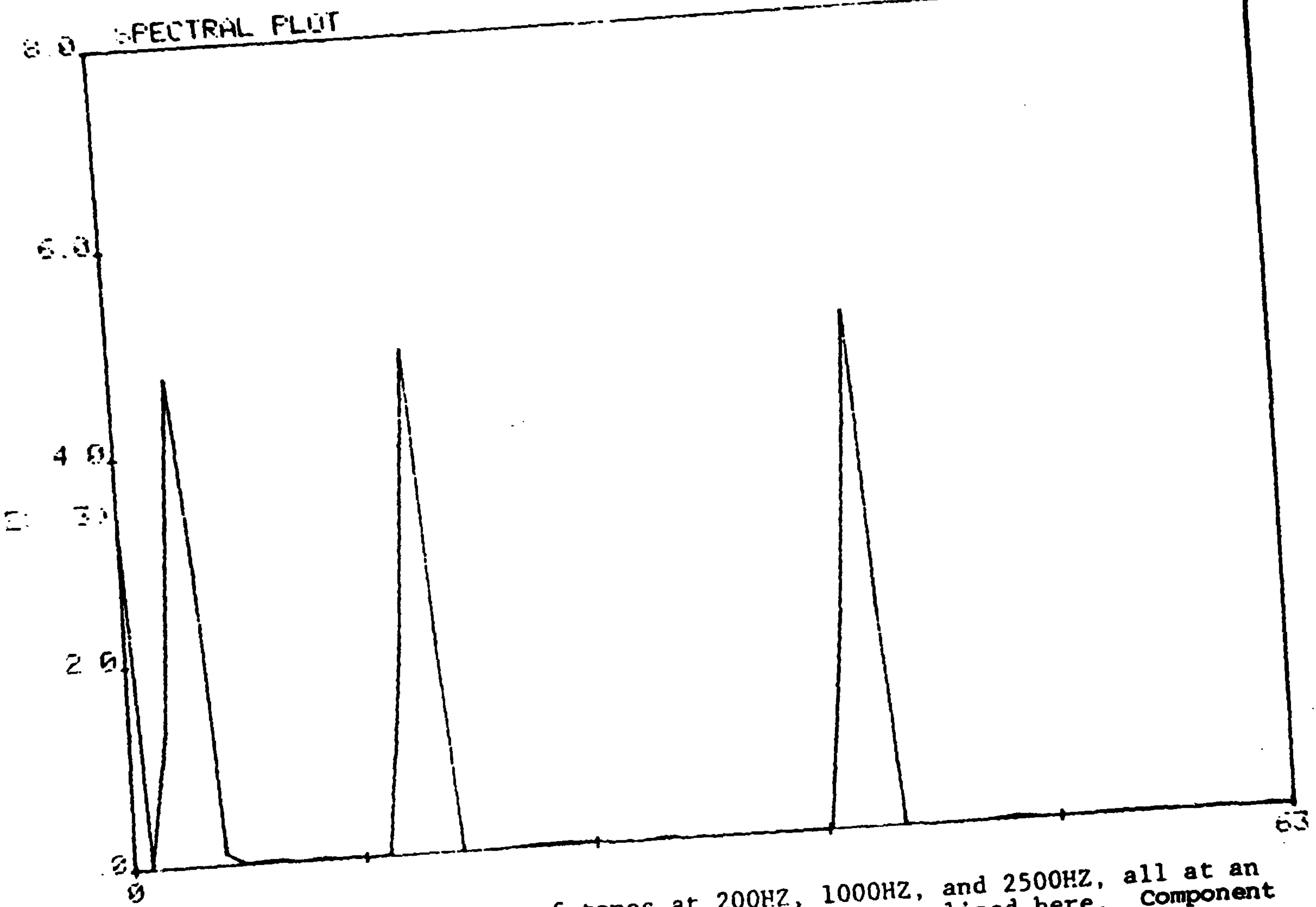


Figure 3 Plot of spectrum of tones at 200HZ, 1000HZ, and 2500HZ, all at an amplitude of 50 units. Observation is normalized here. Component zero is $\sqrt{E_n^1}$, where E_n^1 is observation energy.

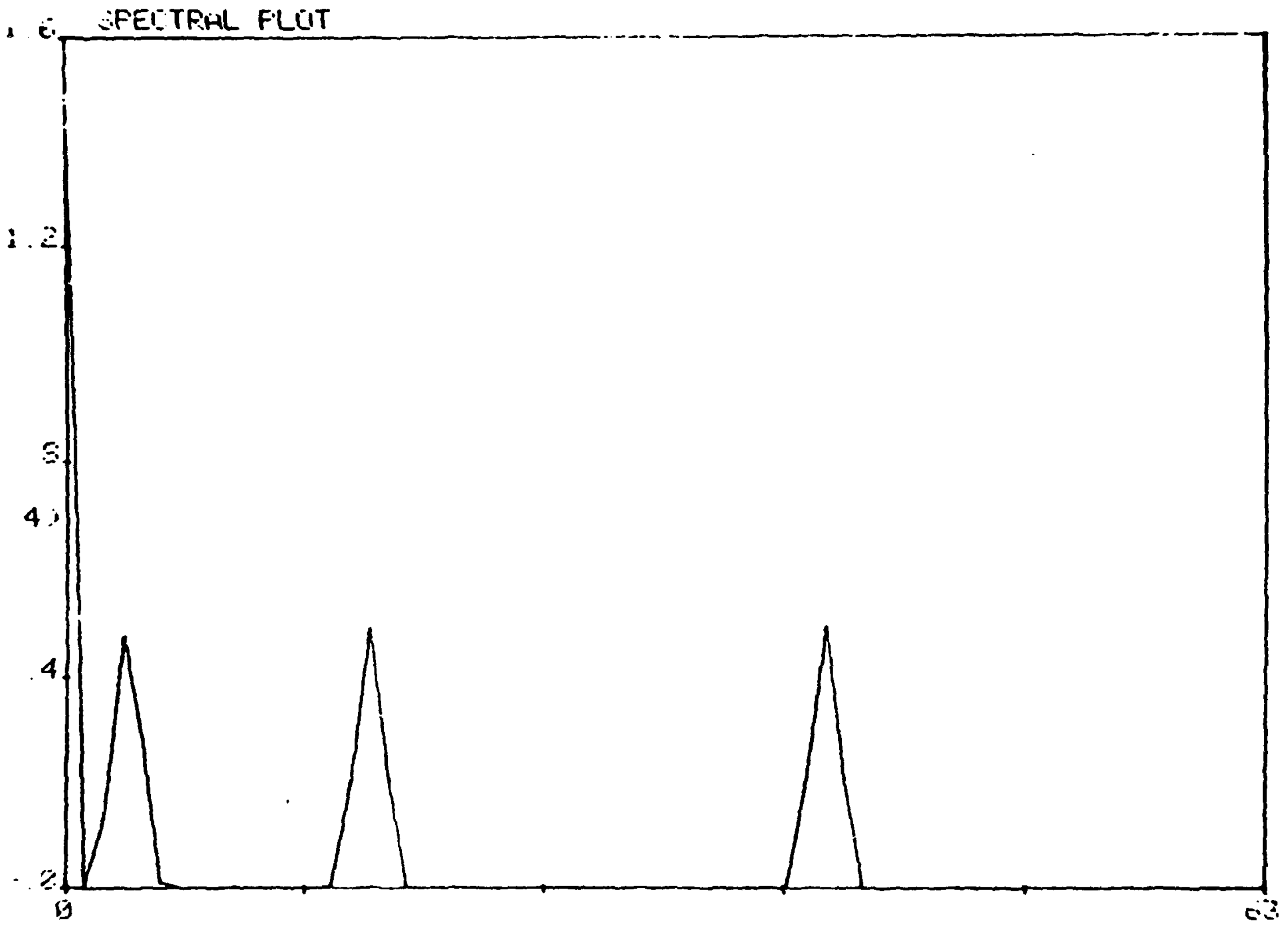


Figure 4 Plot of spectrum of tones at 200HZ, 1000HZ, and 2500HZ; all at an amplitude of 200 units. Observation is normalized here. Component zero is $\sqrt{E_n^1}$, where E_n^1 is observation energy.

are nearly the same in the two observations. The amplitudes of the spectral components differ markedly between the next two figures which were scaled by the factor $10^4/E_n$. Figure 5 was computed from File A which contained tones at an amplitude of 200 units and Figure 6 was computed from File B which contained tones at an amplitude of 50 units. One can also see that energy normalization removes from the observation spectrum all of the variability caused by speech amplitude differences.

Energy normalization has, at least, one other consequence: it limits the maximum Euclidean distance between any two observations. To see this, suppose the observations have been energy normalized by having their components scaled by the factor $K/\sqrt{E}$, where E is the energy in observation being normalized. We write each observation as a vector and note that the energy in the difference between any two observations is:

$$|\underline{x}_1 - \underline{x}_2|^2 \leq |\underline{x}_1|^2 + |\underline{x}_2|^2 \tag{4}$$

Now the energy in each observation is K^2 so the Euclidean distance is:

$$|\underline{x}_1 - \underline{x}_2| \leq \sqrt{2}\ K \tag{5}$$

So the Euclidean distance is upper bounded by $\sqrt{2}$ times the scale factor constant, K.

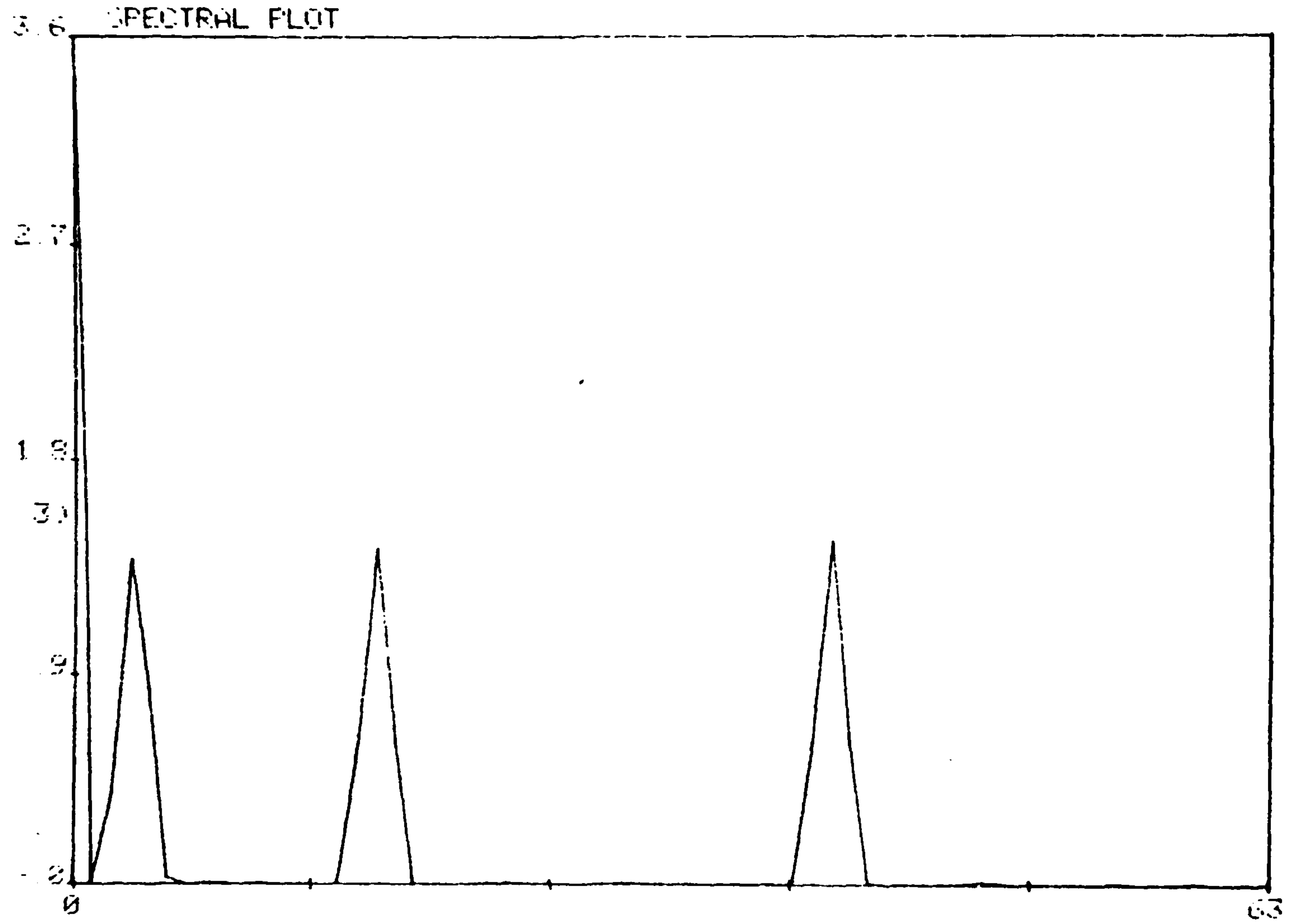


Figure 5 Plot of spectrum of tones at 200HZ, 1000HZ, and 2500HZ, all at an amplitude of 50 units. Observation is divided by its energy. Component zero is $\sqrt{E_n^1}$, where E_n^1 is observation energy.

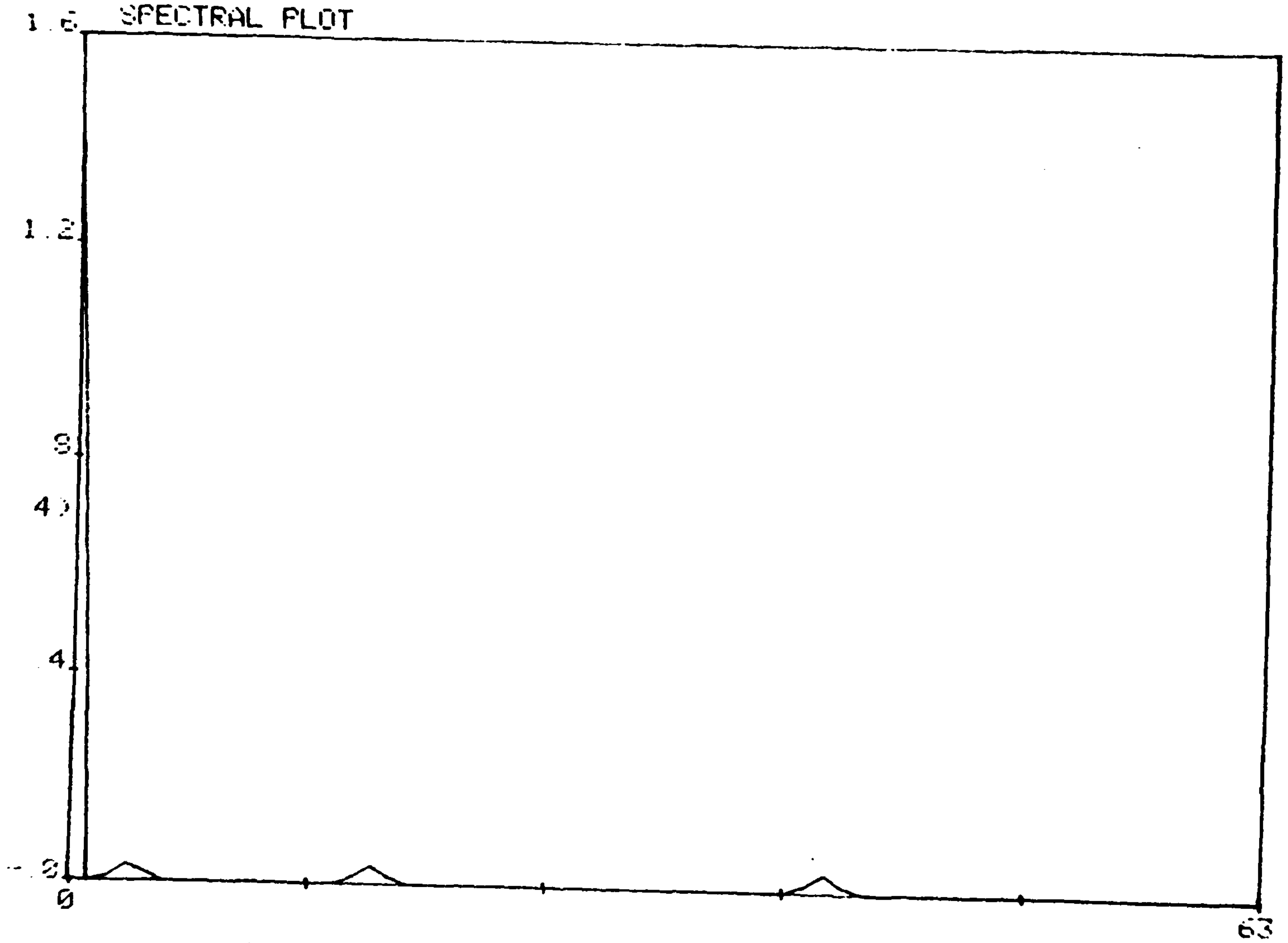


Figure 6   Plot of spectrum of tones at 200HZ, 1000HZ, and 2500HZ, all at an amplitude of 200 units. Observation is divided by its energy. Component zero is $\sqrt{E_n^1}$, where E_n^1 is observation energy.

PHONET GENERATION:

Phonets are selected from a set of observations computed by the Acoustic Analyzer (AA) using the spectrum option. We presently have no systematic means for selecting an adequate set of phonets. Such a set should span the observation space in that any observation can be seen to resemble at least one phonet in a detectable way. It should also be discriminatory in that any observation should resemble only one phonet better than all others to a detectable degree. One last property that an adequate phonet set should have is that of compactness, in that the number of phonets should not be so great as to impose prohibitive computational burdens. Given any phonet set, determining if it has these three basic properties, requires consideration of the distance rule to be used.

DISTANCE COMPUTATION:

Seelandt used the Minkowski One (M1) distance and a smallest distance determination (Ref 1:53-57) to choose five closest phonetic units to each observation run through his distance computer. He used this measure because it was convenient to compute. Our Acoustic Analyzer includes this distance measure as one option. We also included the Euclidean, or Minkowski Two (M2) distance (Ref 9:232). One option of our Acoustic Analyzer is a distance computer which uses either the M1 or M2 measure to calculate a symmetric distance matrix, φ. The element $\varphi(i, j)$ is the distance between the i^{th} observation and the j^{th} phonet. The other distance option in our AA uses either distance measure to choose a selectable number of closest phonets to each observation. In the file for each observation is the observation number as the first element, the observation energy as the second element, an ordered pair of (phonet number, corresponding maximum distance), in the next two elements and the selected number of ordered pairs (phonet number, corresponding minimum distance). These last ordered pairs are themselves ordered minimum distance choice-to-maximum. Files for consecutive observations are contiguously arranged in the output file.

We chose to implement the M2 distance measure because it is the Euclidean distance in a geometrical sense in the absence of noise. We implemented the M1 rule to provide the capability to compare results obtained using the Acoustic Analyzer with those using Seelandt's M1 implementation. It is acknowledged that other distance measures and more elaborate decision rules will be needed to achieve a specified error probability in noise. This implementation assumes a completely deterministic system without noise of any kind. Noise is introduced into the analysis of the feature space cluster algorithm described in the next chapter.

III. RESULTS.

THE ACOUSTIC ANALYZER (AA):

One objective of this project was to implement a machine like Seelandt's (Ref 1) Speech Sound Analysis Machine (SSAM) on the Eclipse AP/130 Array Processor, taking advantage of its fast computational capability. This work resulted in delivery of a fast, flexible software package with the required features:

(1) A flexible spectral computing option which transforms time slices of speech into observations.

(2) A flexible distance computer which calculates a symmetric distance matrix between each observation and phonet.

(3) Another distance computer which chooses a selectable number of best matches from a distance matrix.

(4) A graphics capability which allows one to view plots of speech files, observation files, phonet files, and distance files.

The ability to select parameters such as those which define the FFT time window, define processing to be done on the transformed spectral units, and characterize the detection scheme, has been included in our Acoustic Analyzer. We chose to implement a 128 point, overlapped FFT window.

In the discussion on observation generation, we saw that normalizing the energy in each observation removed much of the variability from speech amplitude variation. In the next section, we will illustrate effects of other parameter selections.

PARAMETER VARIATION:

In this section, we illustrate observations and distances computed using several parameter settings. We use only one speech file, CT56.OB, which is an adult male saying the words "five" and "six." It is beyond the scope of this project to illustrate parameter variation effects for a large number of speech files; but, there is reason to believe that such effects would be similar to those illustrated here.

Seelandt (Ref 1) identified the following parameters as requiring study in order to determine optimum settings:

(1) The number of consecutive time slices grouped together to represent a phoneme. Seelandt's phonetic units were five time slices long.

(2) The FFT window size. Seelandt used a 64 point window.

(3) Rate at which the speech time signal is sampled when digitized. Seelandt used speech sampled at 8KHz.

(4) Window shape: rectangular, Hamming, or other. Seelandt used the Hamming window.

(5) Spectral shaping: pre-emphasis to emphasize frequency components above a corner frequency, and de-emphasis to attenuate frequency components below a corner frequency.

(6) Spectral domain thresholding. Seelandt normalized the energy in the spectrum in each time slice when that energy was above a threshold. He attenuated the spectrum in time slices which contained less energy than the threshold.

PHONEME LENGTH:

We decided to treat phonetic units one time slice long, rather than form five time slice phonemes, for the following reasons. First, it is more convenient to process single unit phonets than multiple unit phonemes. Second, the dynamic behavior of speech segments would be visible with more resolution in a plot of distance-versus-phonet number. Third, until the dynamic behavior of adjacent short-time speech units is better characterized, we believed that it would not be prudent to average their effect in a distance measure. We reasoned that it would be better to preserve the time slice-by-time slice variation. Then when short-time speech dynamics are better understood, one can simply reformat spectral and distance files generated by our Acoustic Analyzer to accommodate any desired phoneme length.

Computing distances on single time slice phonets, we found that adjacent time slices were similar in the midst of a vowel and not similar in unvoiced fricatives. From a spectrogram of file CT56.OB, we knew that the speaker was uttering the word "five" during time slices 40-80 and the "si" part of "six" during time slices 90-100. The spectrogram is provided in Appendix A. We used a 128 point, non-overlapped Hamming window with 10dB/Octave de-emphasis ending at 300Hz and 10dB/Octave pre-emphasis starting at 500Hz. We computed the M2 distance between the file and itself for time slices 41-102; that is, we ran observations 41-102 against phonets 41-102 of the same file. Plots of observations 41, 51, 61, 71, 81, and 91 against phonets 41-102 are shown in Figures 8, 10, 12, 13, 14, and 15, respectively. The similarity of adjacent observations in the vowel regions in Figures 10 and 12 are apparent, as is the dissimilarity of adjacent observations elsewhere. This effect is illustrated on an expanded scale in Figures 9 and 11, which show distances between the observations 45 and 55, respectively, plotted against phonets 41-70.

We have seen this effect every time we have looked for it. Adjacent time slices resemble each other in both M2 and M1 space much more in regions of vowels when a spectrogram shows clear formant structure than in regions where a spectrogram shows little formant structure. This formant structure is visible in plots of the observations.

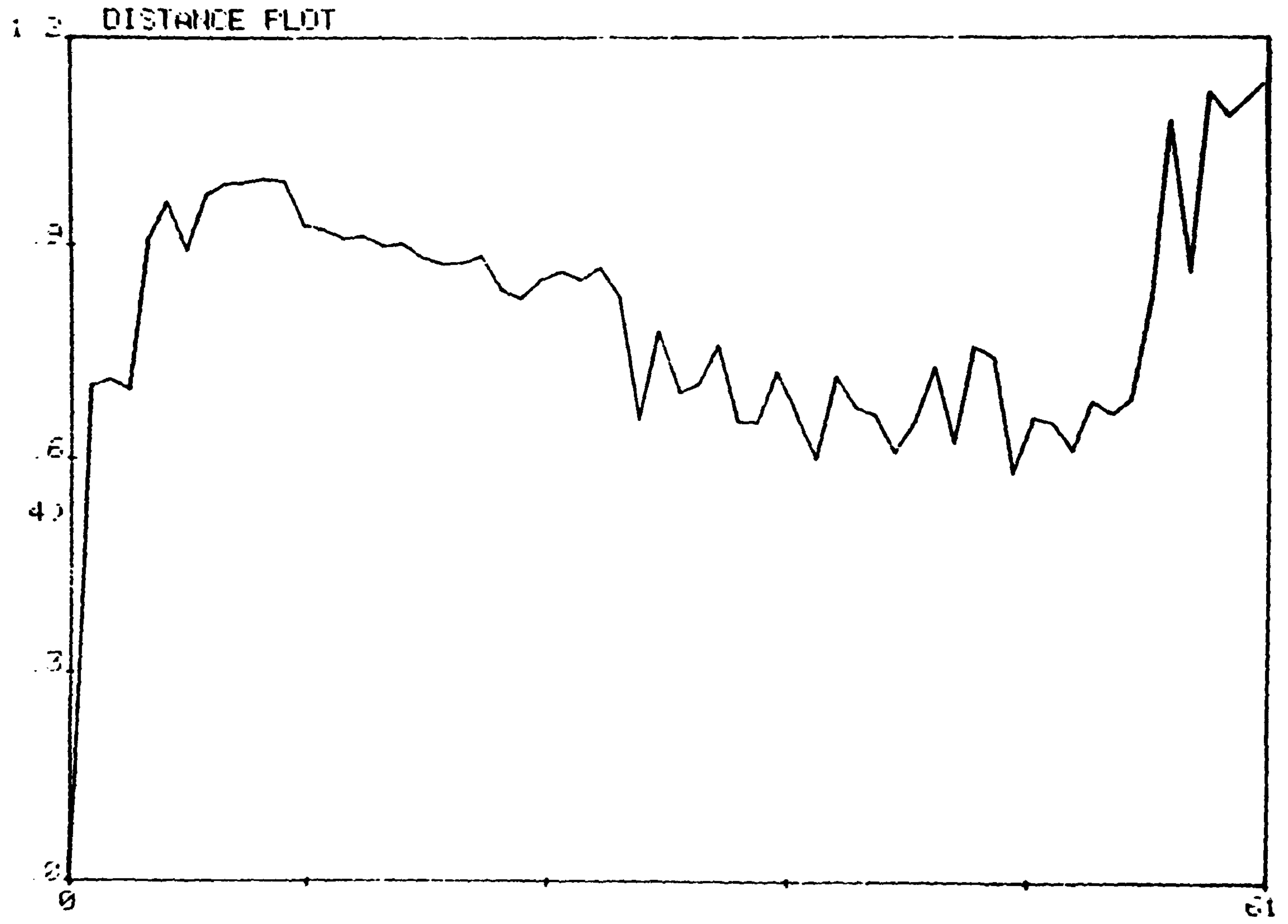


FIGURE 8. Distance plot of observation number 41 against 41 through 102. M2 rule, 128 point, nonoverlapped Hamming window, energy normalized.

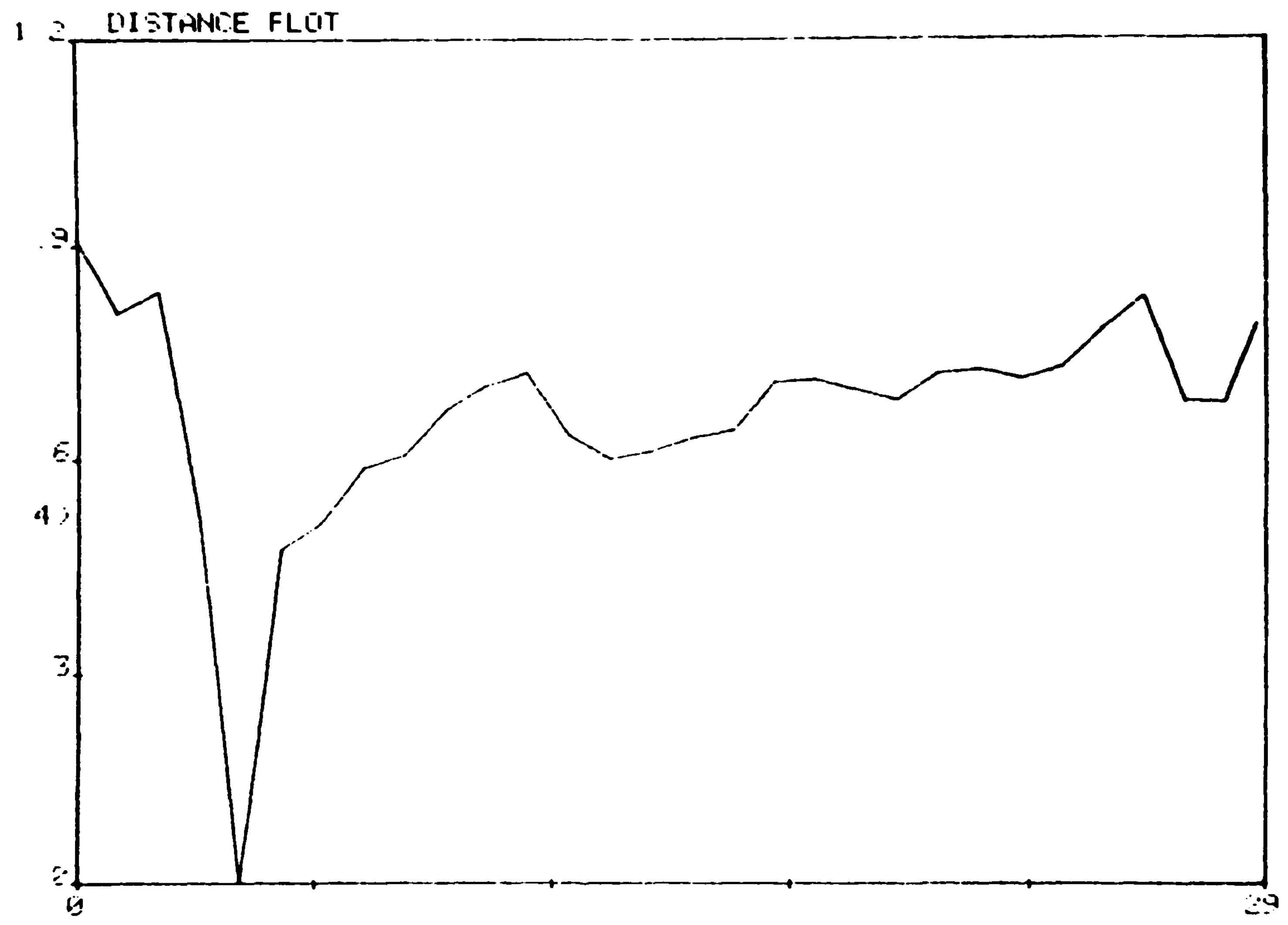


FIGURE 9. Distance plot of observation number 45 against 41 through 70. M2 rule, 128 point, nonoverlapped Hamming window, energy normalized.

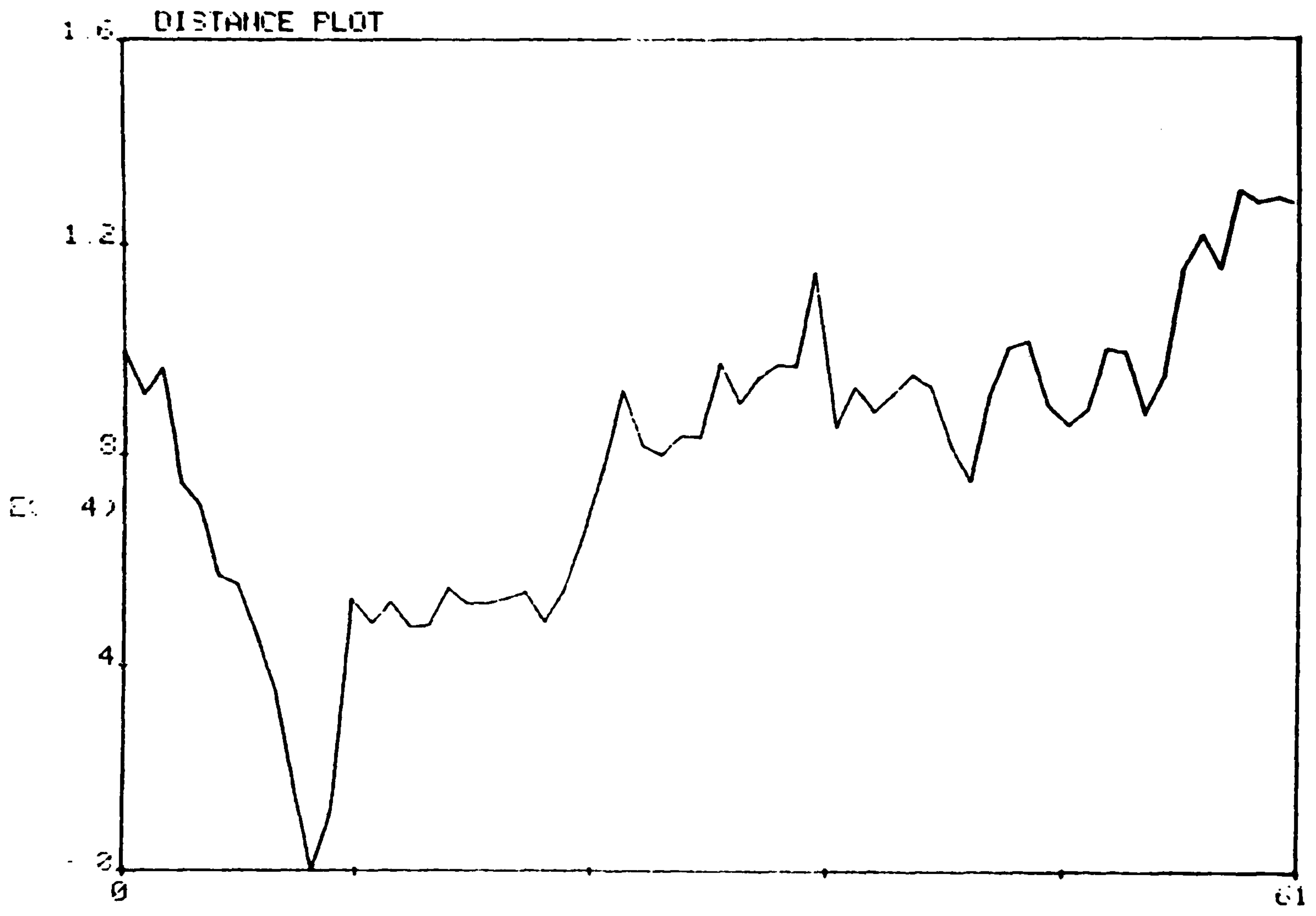


FIGURE 10. Distance plot of observation number 51 against 41 through 102. M2 rule, 128 point, nonoverlapped Hamming window, energy normalized.

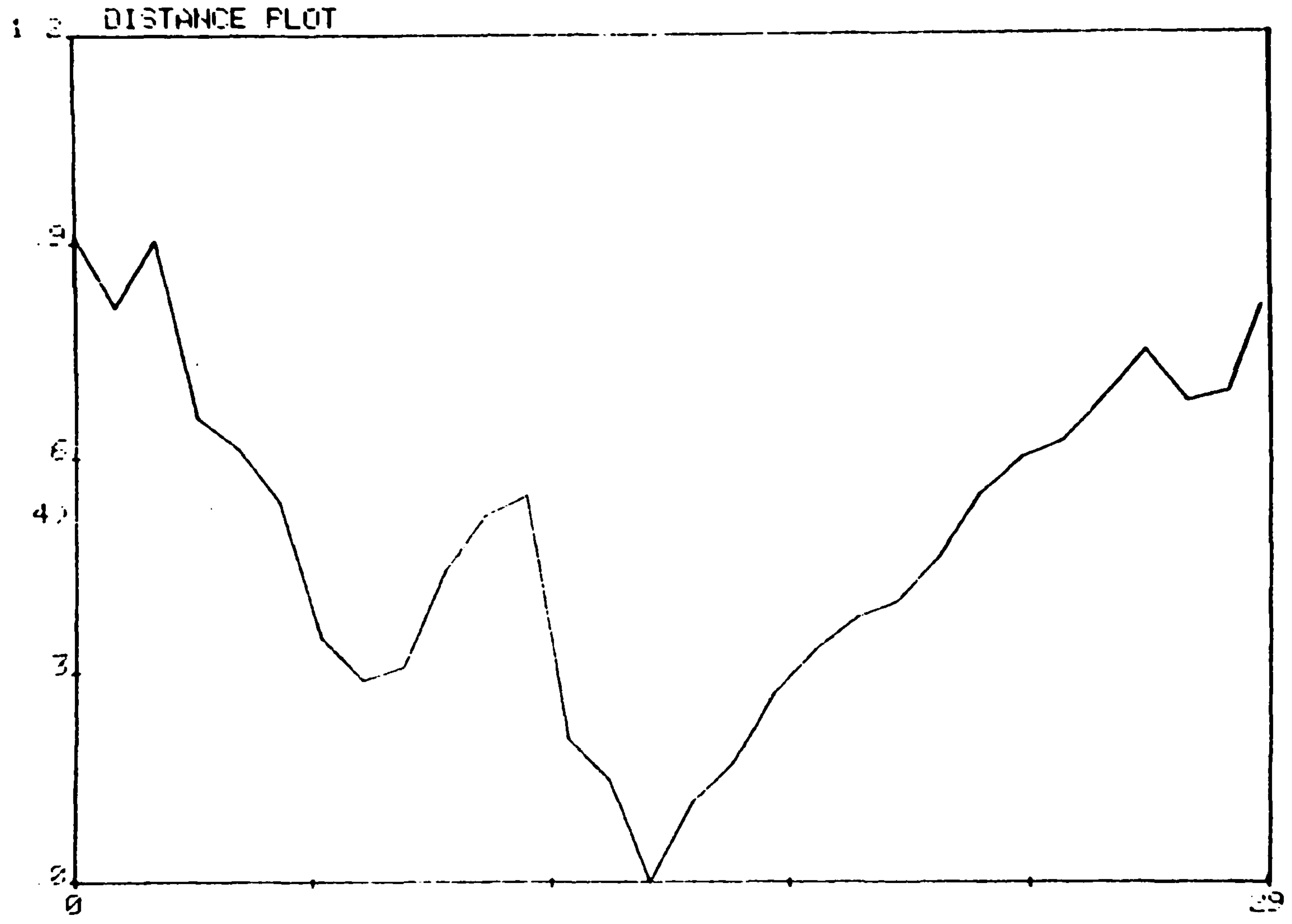


FIGURE 11. Distance plot of observation 55 against 41 through 70. M2 rule, 128 point, nonoverlapped Hamming window, energy normalized.

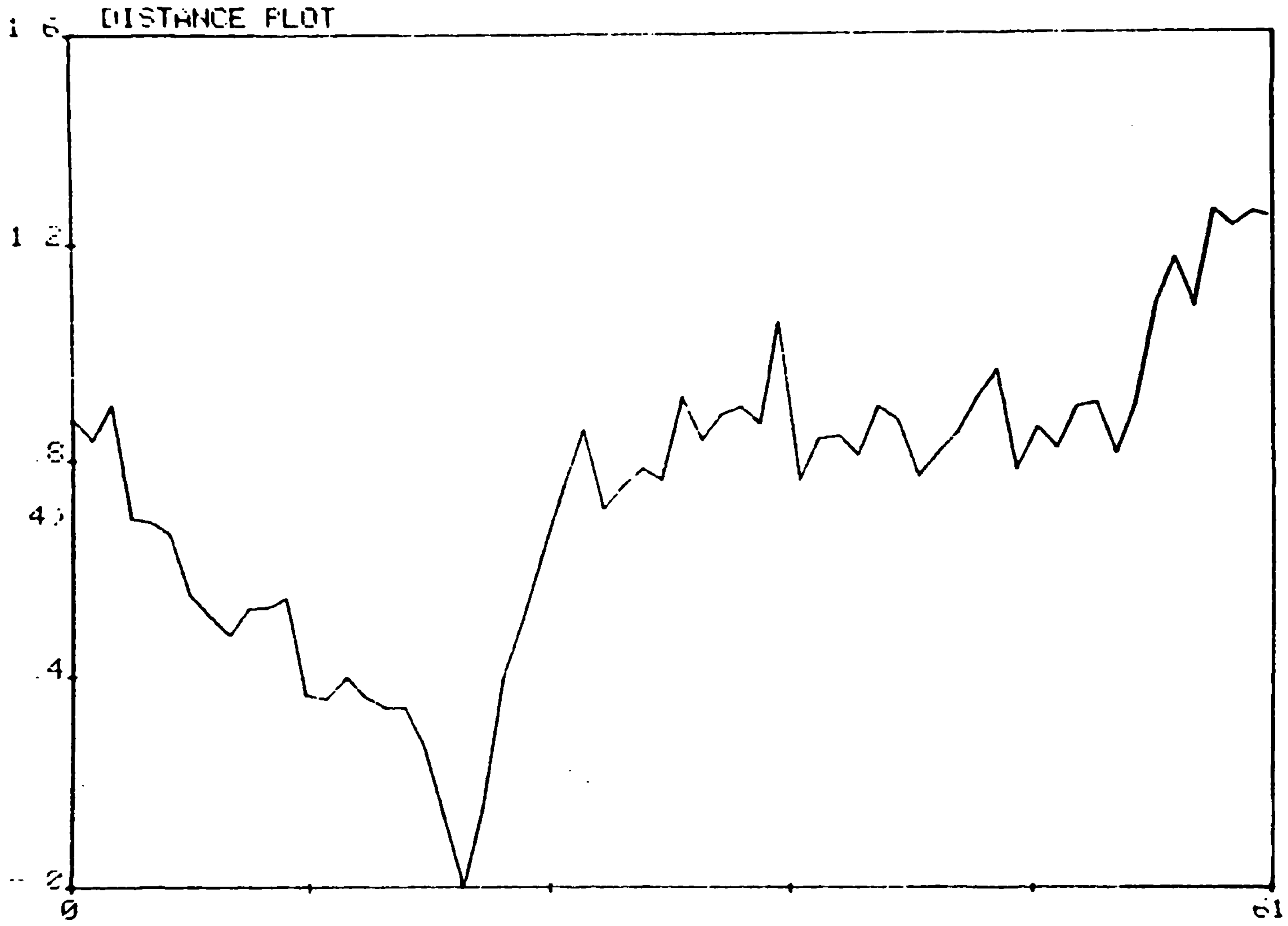


FIGURE 12. Distance plot of observation 61 against 41 through 102. M2 rule, 128 point, nonoverlapped Hamming window, energy normalized.

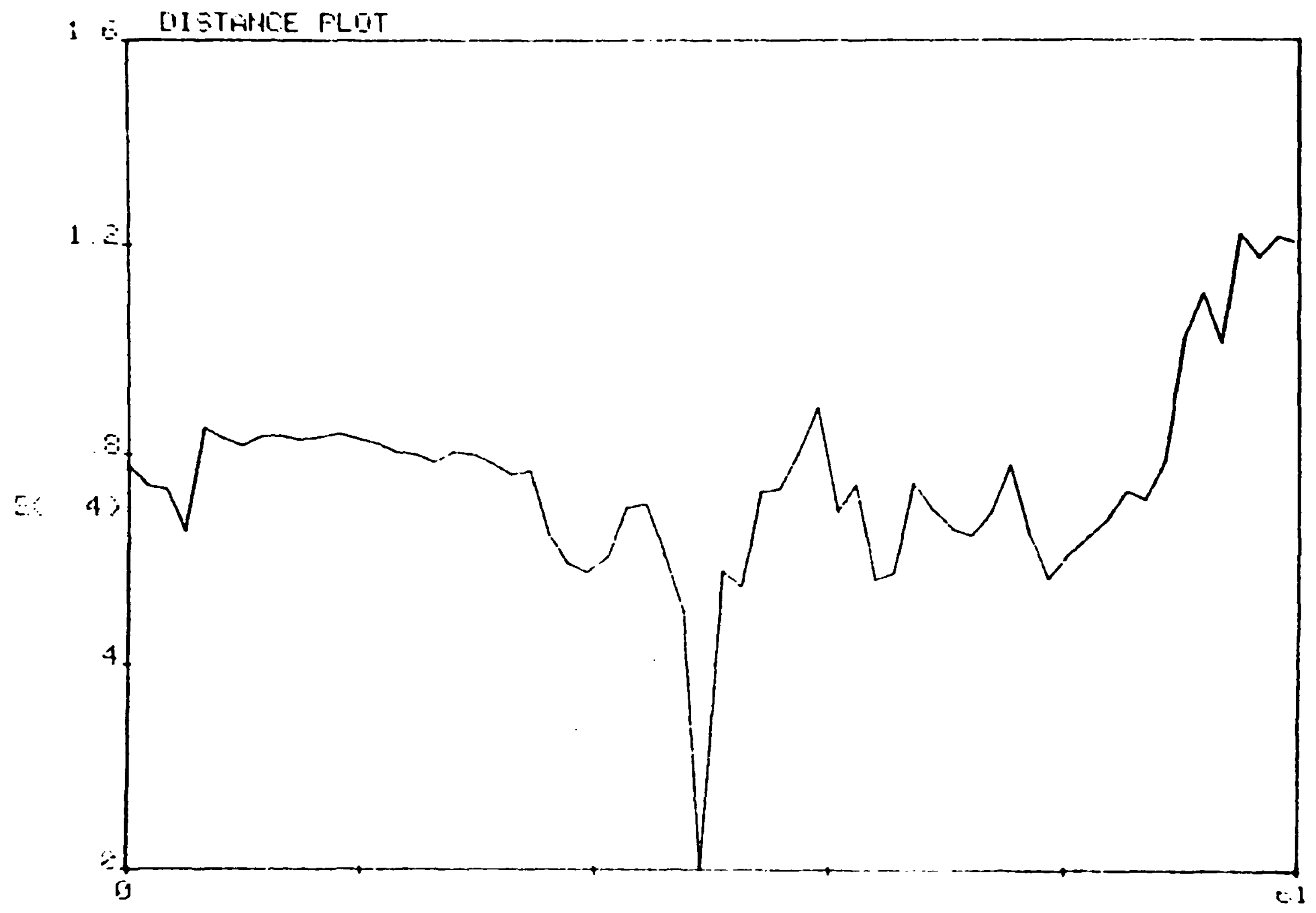
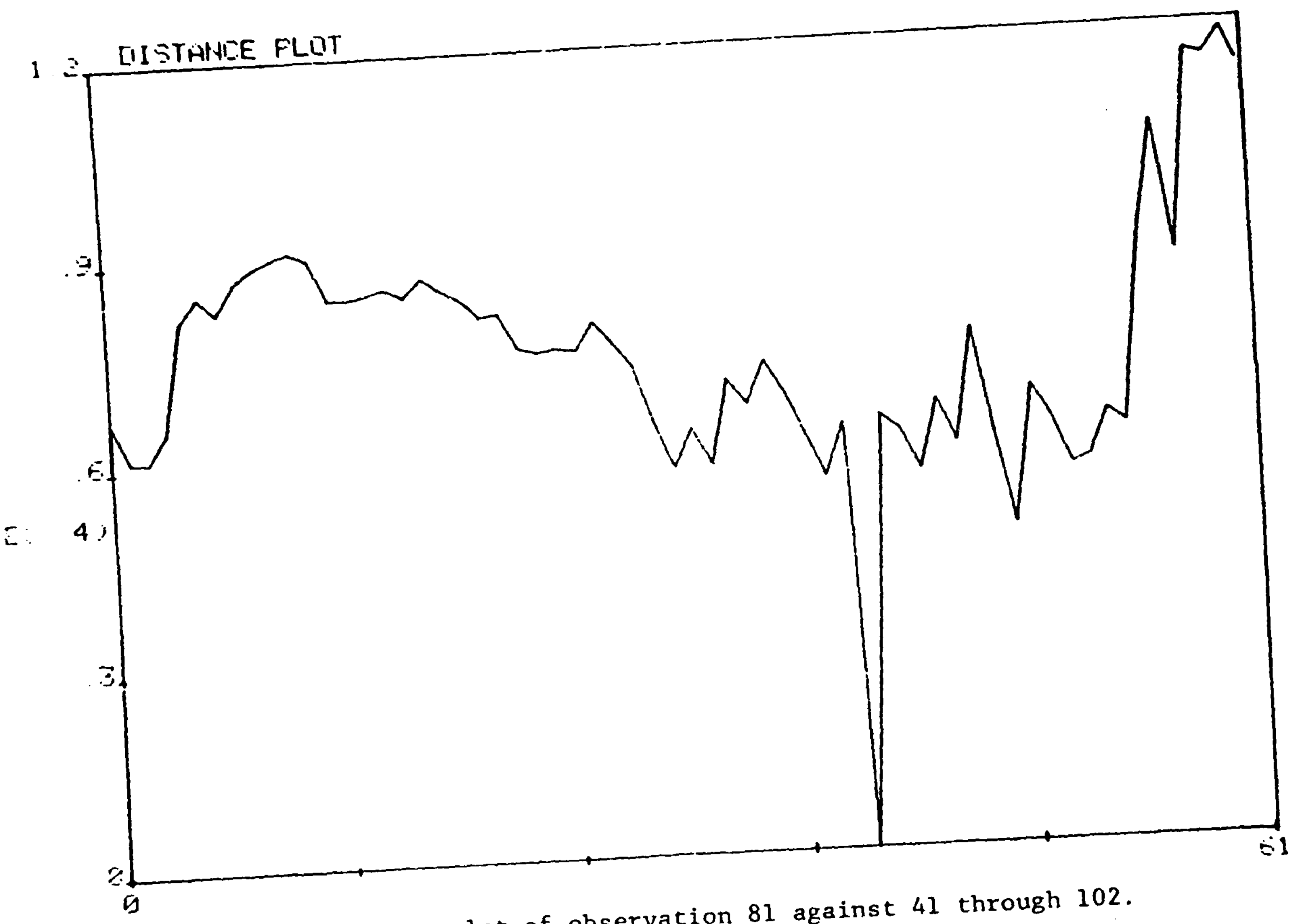


FIGURE 13. Distance plot of observation 71 against 41 through 102. M2 rule, 128 point, nonoverlapped Hamming window, energy normalized.



FIGURF 14. Distance plot of observation 81 against 41 through 102. M2 rule, 128 point, nonoverlapped Hamming window, energy normalized.

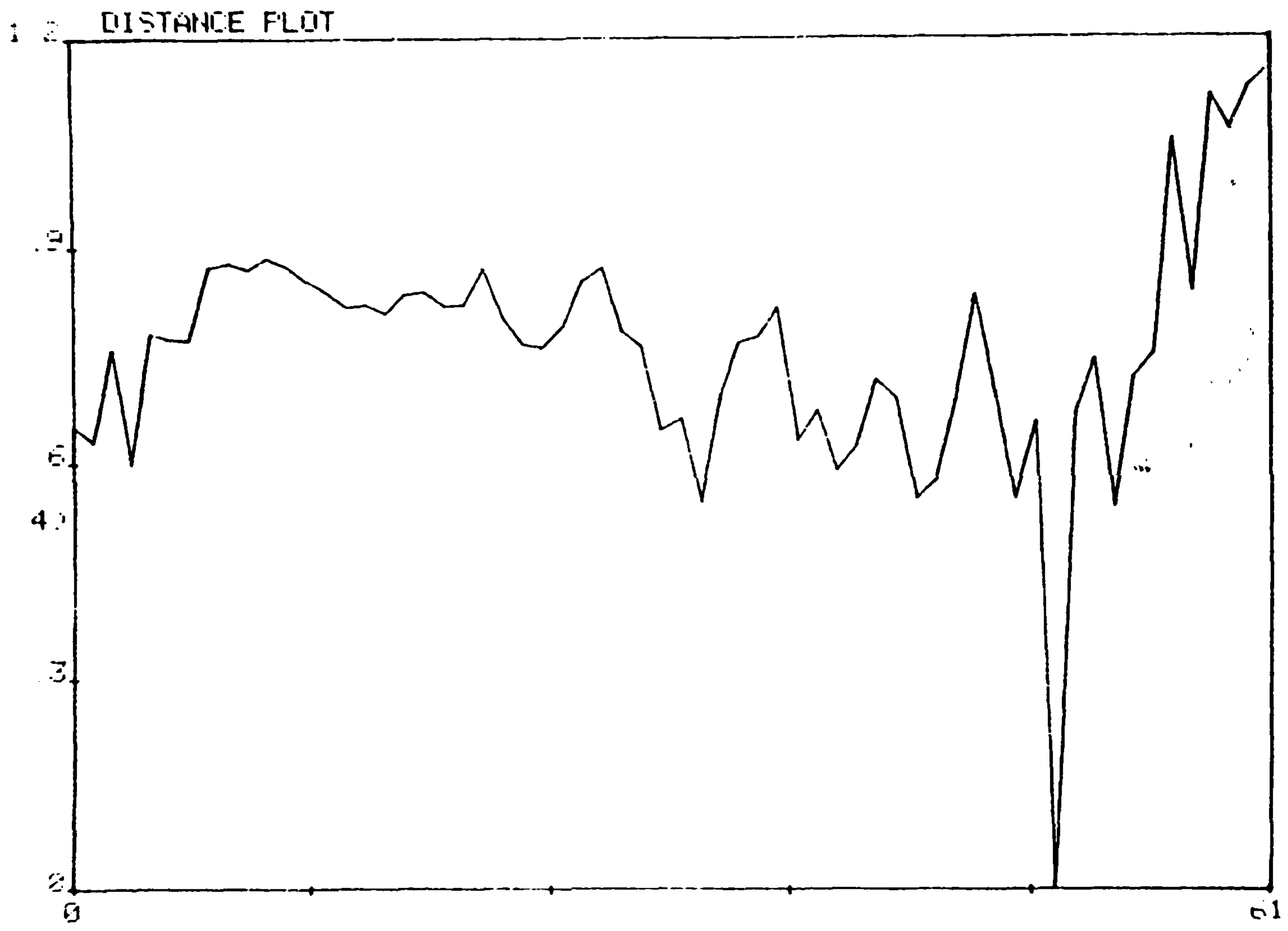


FIGURE 15. Distance plot of observation 91 against 41 through 102. M2 rule, 128 point, nonoverlapped Hamming window, energy normalized.

Figures 16 and 17 are plotted using the spectral plot option of our Acoustic Analyzer, and are observations 45 and 47 from the observation file above. Though these observations are separated by 32MSEC, their resemblance is clear.

FFT WINDOW SIZE:

We have discussed our choice of FFT window size in Section II of this report. A 128 point window provides more frequency resolution than does a 64 point window, and leads to useful behavior of short-time energy with regard to voiced/unvoiced speech (Ref 8). We also found when we computed distances between an observation file and itself in voiced vowel regions of pronounced formant structure, that adjacent observations had similar formant structure using a 64 point window as with a window size of 128 points. To illustrate, observations were computed using a 64 point Hamming window. Observations were de-emphasized at 10dB/Octave below 300Hz and pre-emphasized at 10dB/Octave above 500Hz. The formant structure can be seen to develop in the sequence of plots of observation number 86, 87, 88, 89, and 90; Figures 18, 19, 20, 21, and 22, respectively. These observations are in the same speech segment as Figures 8-17; there are twice the number of observations in a time segment, here, at 64 points per time slice. The M2 distance measure can be seen to develop broad valleys in

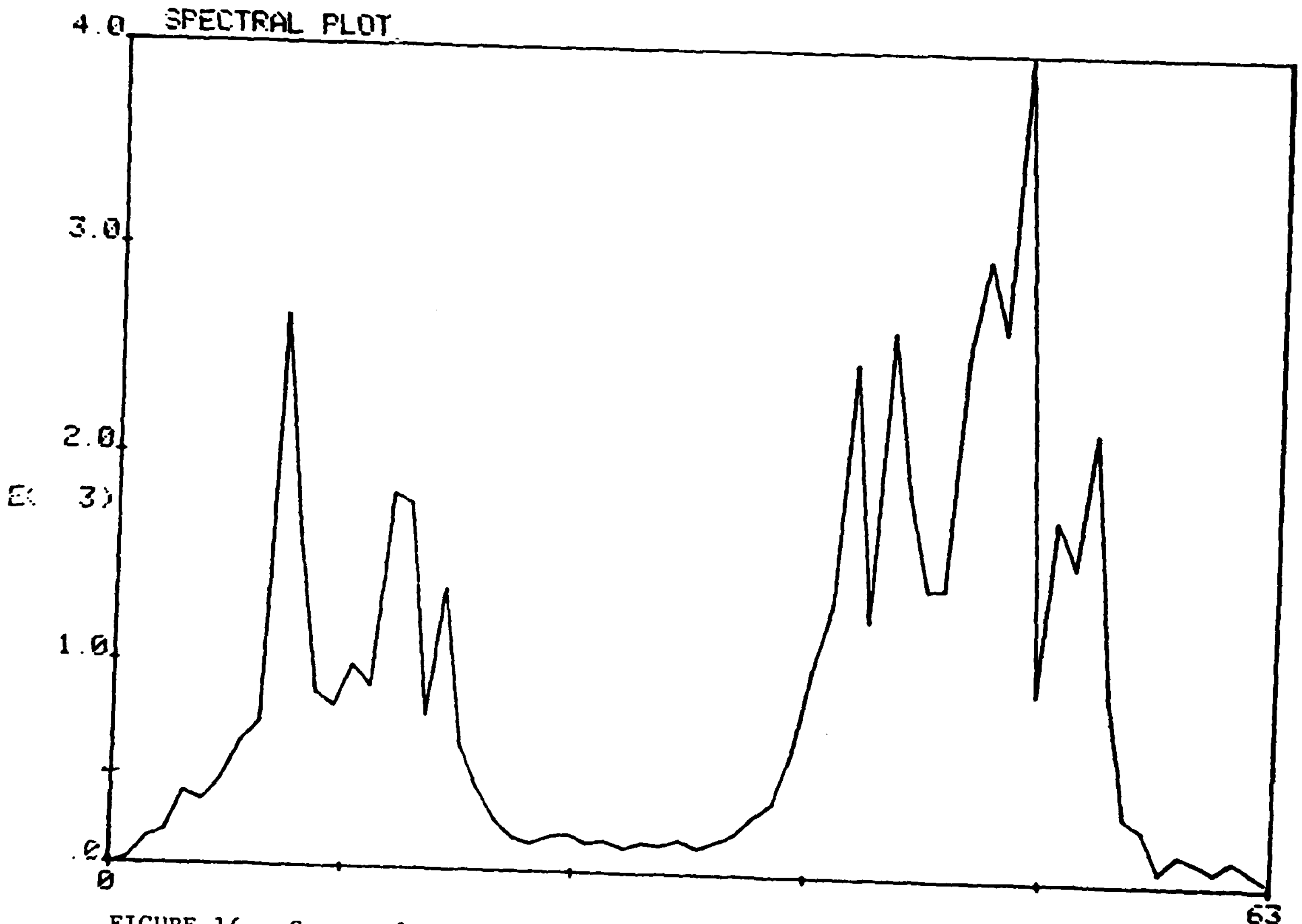


FIGURE 16. Spectral plot of observation number 45. One-hundred twenty-eight (128) point, nonoverlapped Hamming window, energy normalized.

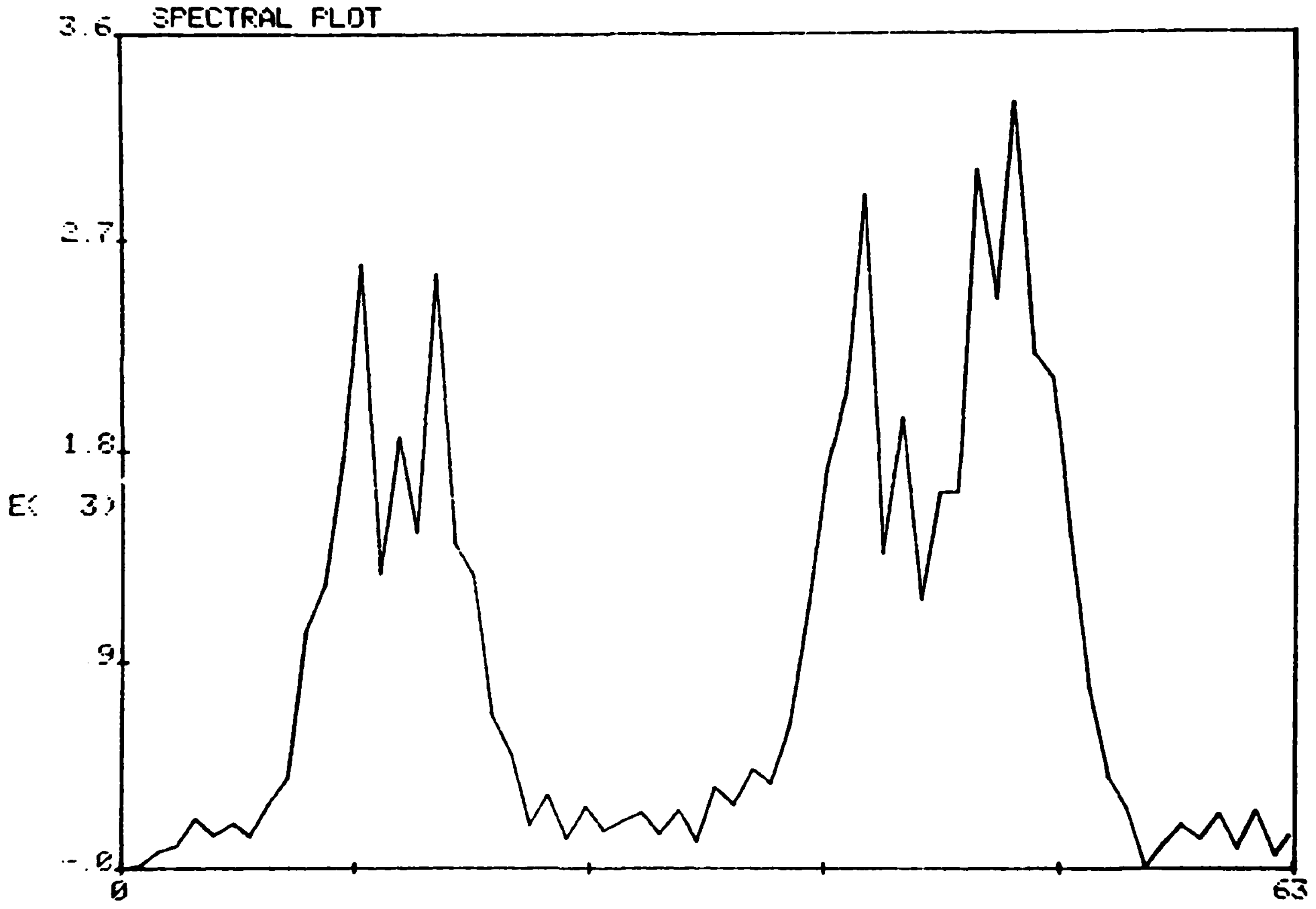


FIGURE 17. Spectral plot of observation number 47. One-hundred twenty-eight (128) point, nonoverlapped Hamming window, energy normalized.

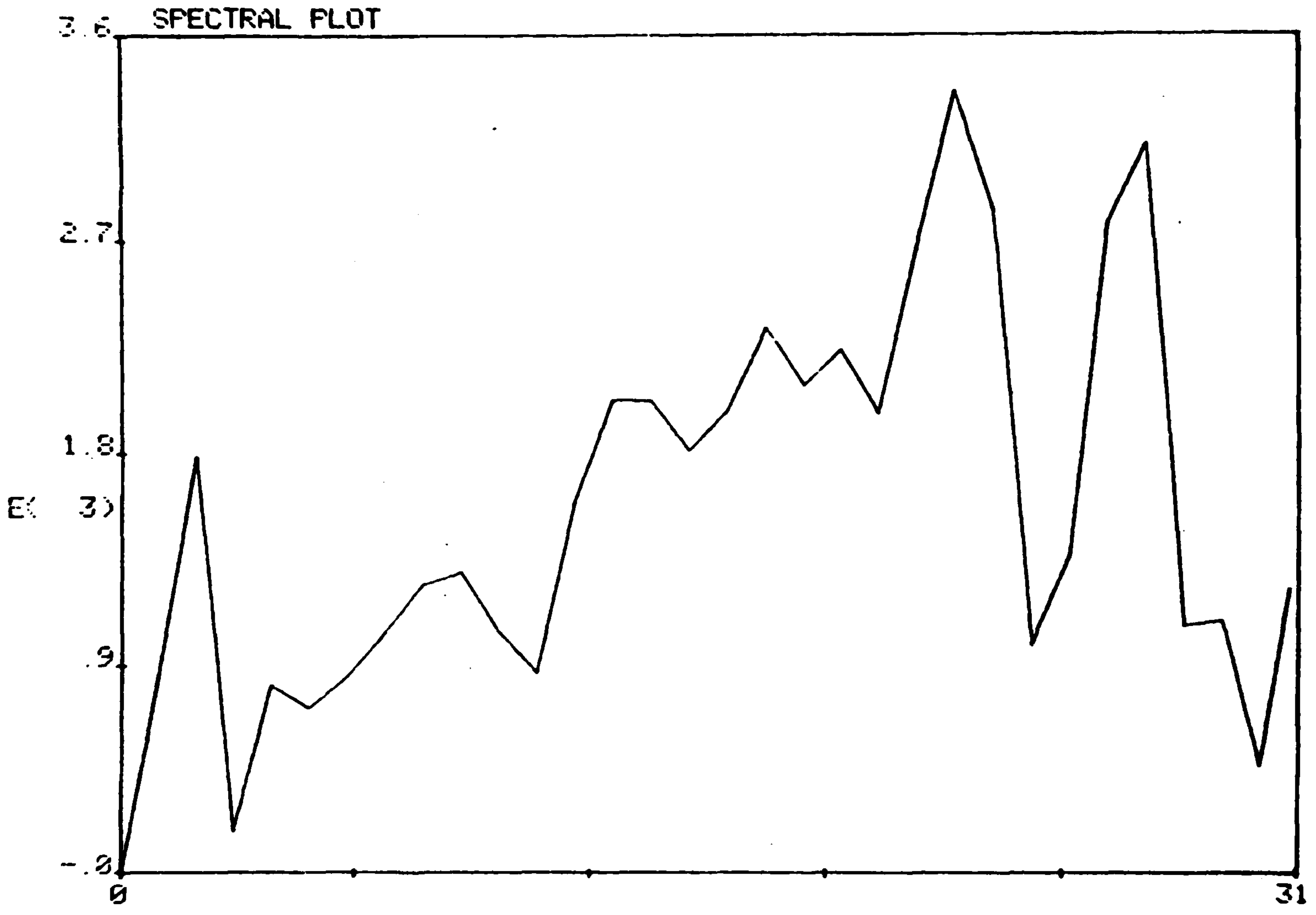


FIGURE 18. Spectral plot of observation number 86. Sixty-four (64) point Hamming window, energy normalized.

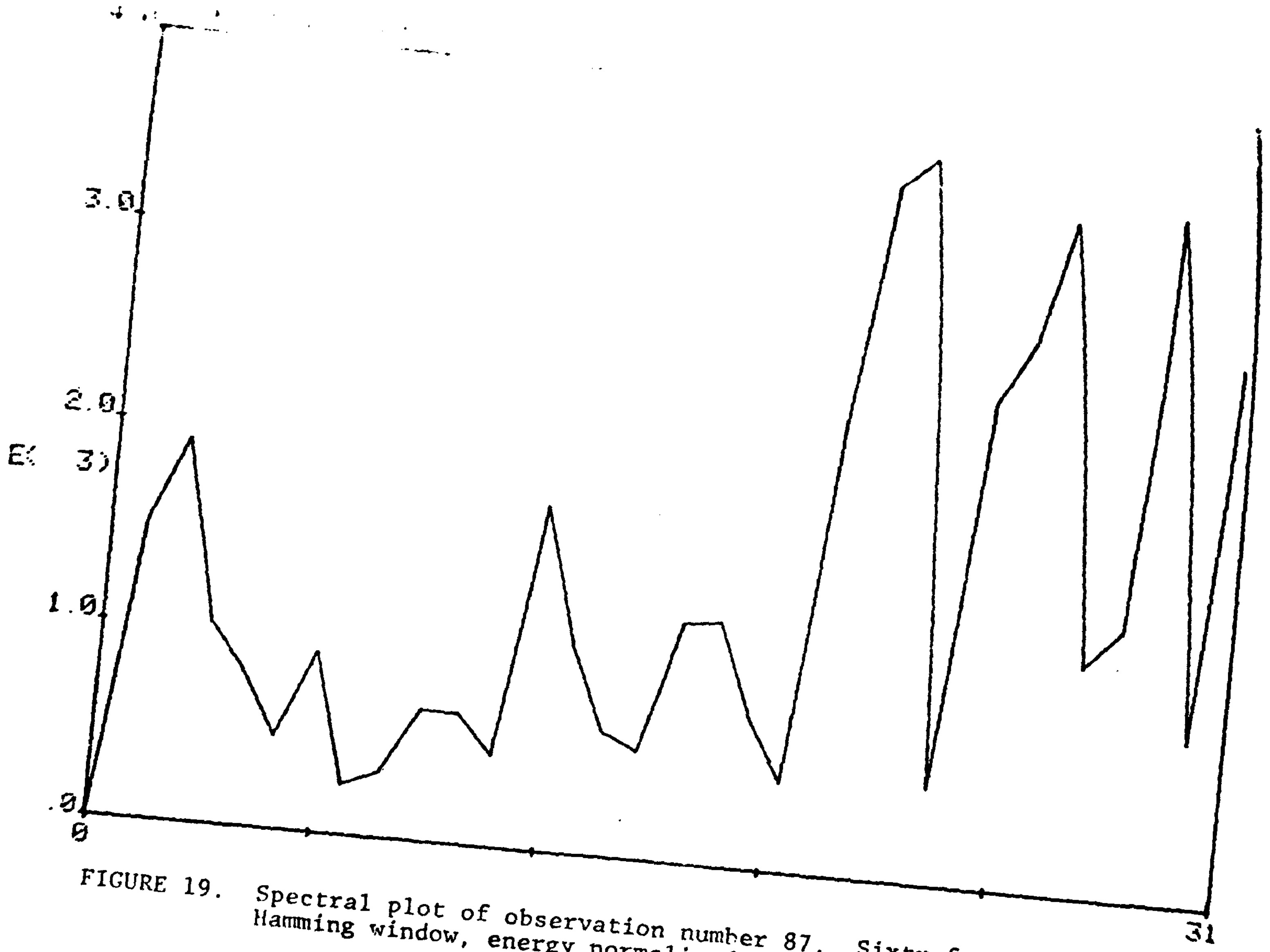


FIGURE 19. Spectral plot of observation number 87. Sixty-four (64) point Hamming window, energy normalized.

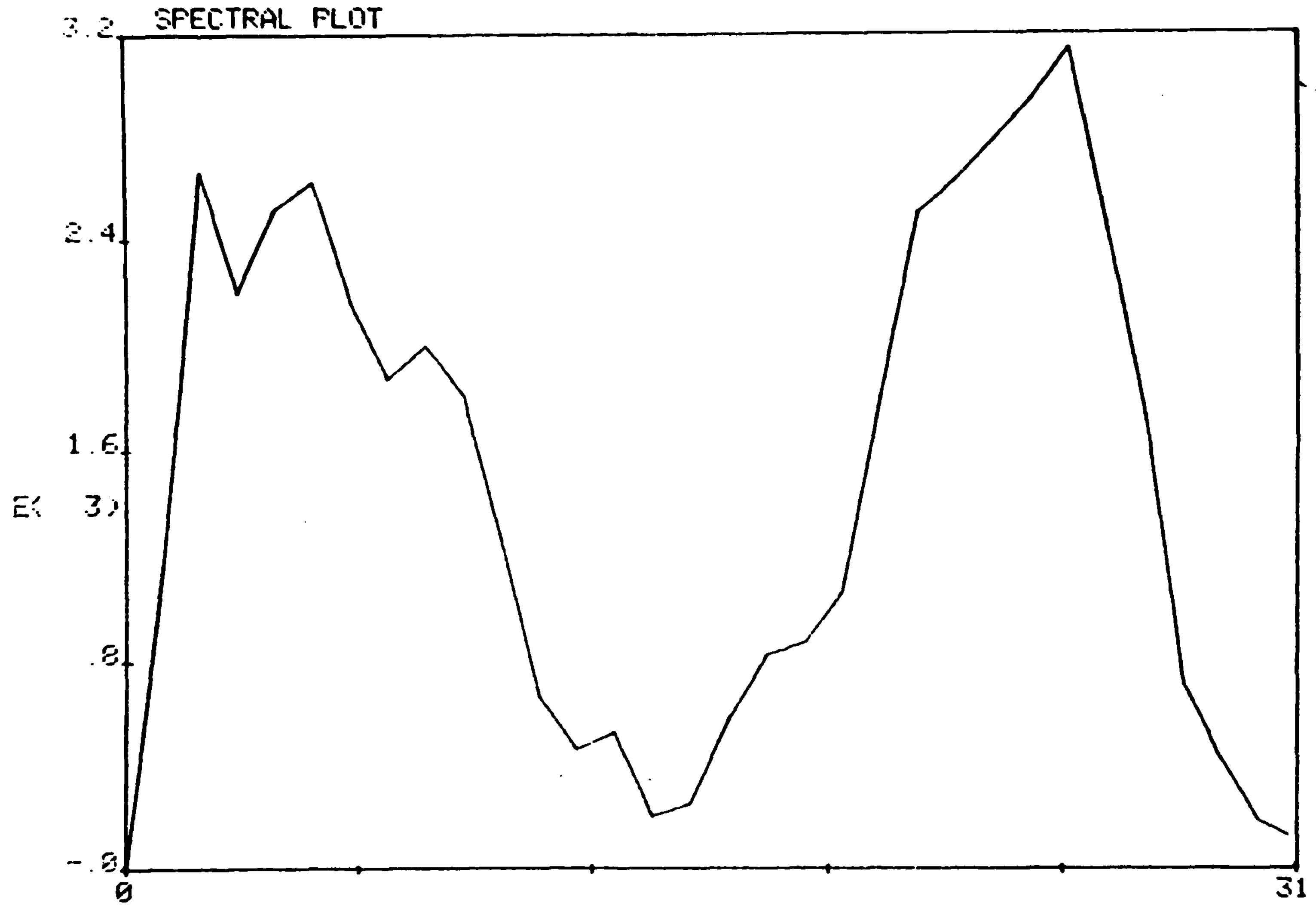


FIGURE 20. Spectral plot of observation number 88. Sixty-four (64) point Hamming window, energy normalized.

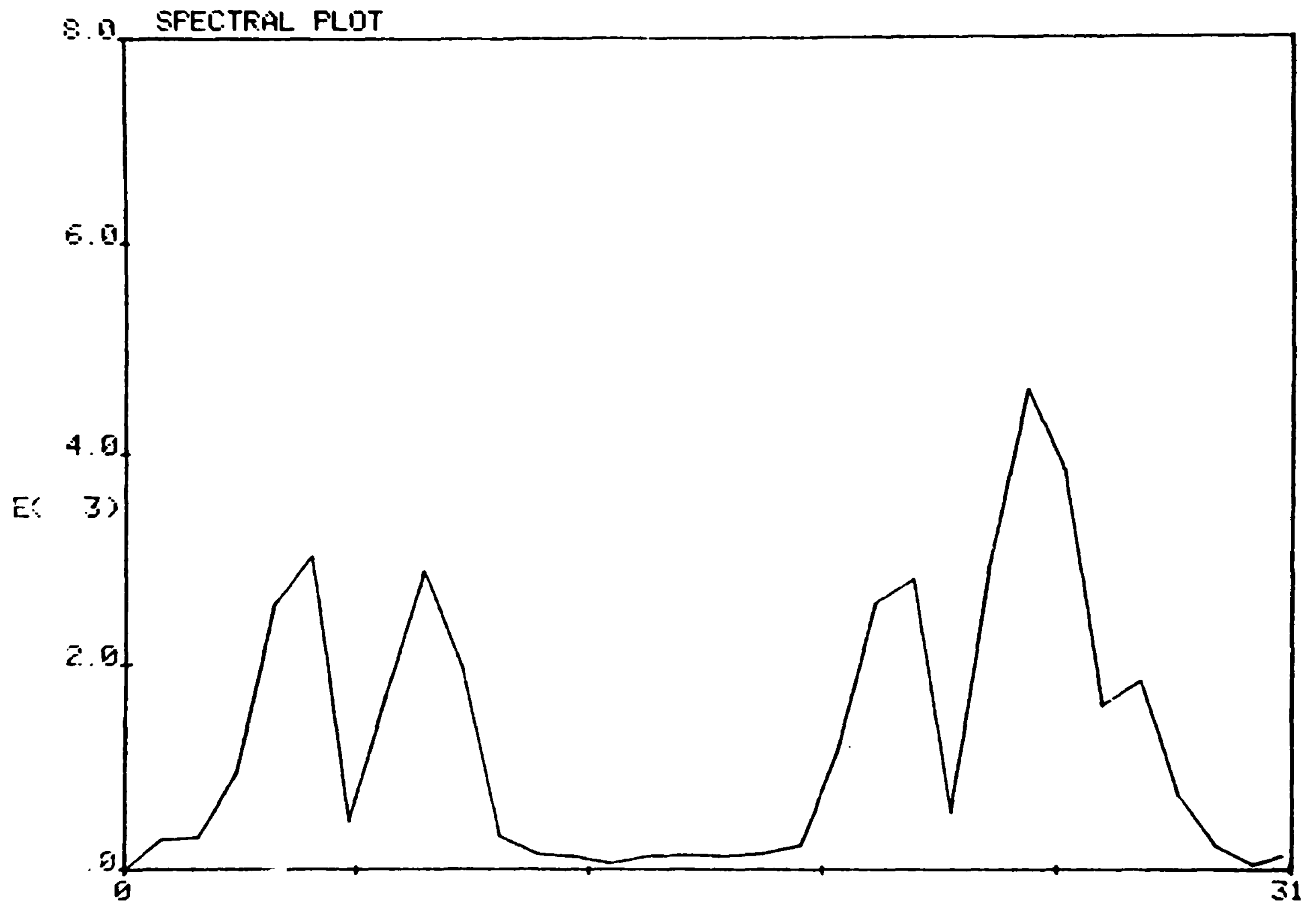


FIGURE 21. Spectral plot of observation number 89. Sixty-four (64) point Hamming window, energy normalized.

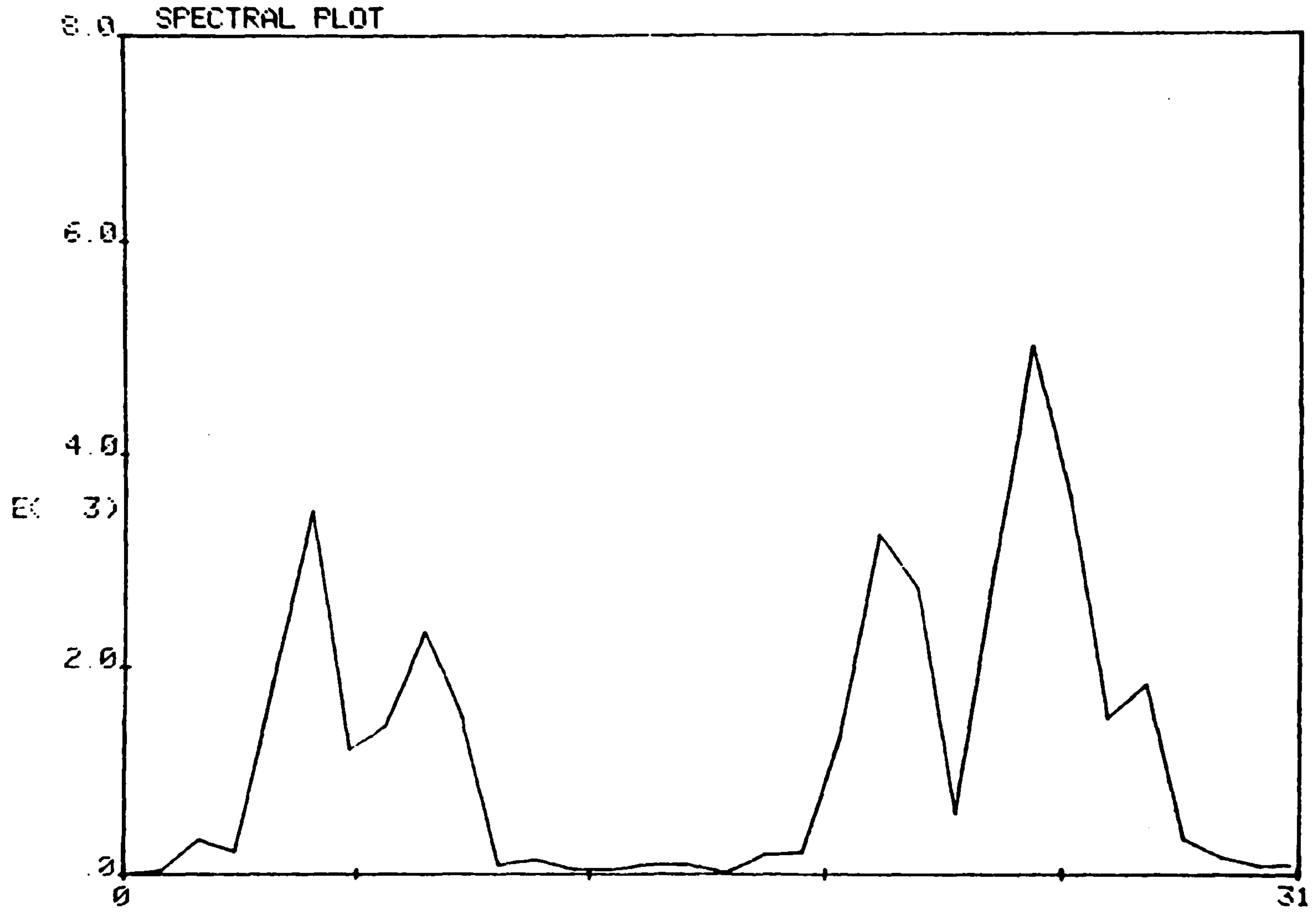


FIGURE 22. Spectral plot of observation number 90. Sixty-four (64) point Hamming window, energy normalized.

regions of most pronounced formant structure in Figures 23, 24, 25, 26, 27, and 28. These are plots of observations number 85, 90, 115, 120, 125, and 130, respectively, all against phonet numbers 81-144.

SAMPLING RATE, WINDOW SHAPE, AND EMPHASIS:

We used speech files digitized at an 8KHz sampling rate and 12 bit quantization because it was the highest sampling rate available. It is well known, of course, that speech sampled at this rate is perfectly intelligible when reconstructed. Seelandt describes the digitizing aparatus (Ref 1). Here we discuss the advisability of changing that rate. We also discuss window shape.

The capability or our Acoustic Analyzer to calculate FFTs of window size 512 or 1024 points allows one to double or quadruple the sampling rate while maintaining the behavior of short-time energy discussed by Rabiner and Schafer (Ref 8) and in Section II of this report. Using the Array Processor, the computational burden of large windows is light. The advantages would not only be an increase in spectral resolution, but also a decrease in spectral distortion due to the longer window. The effects of window shape become more important as pre-emphasis is applied to high frequencies, especially so when a large portion of the energy in speech is concentrated in the lower frequency range. In this case, the sidelobes from the low frequency

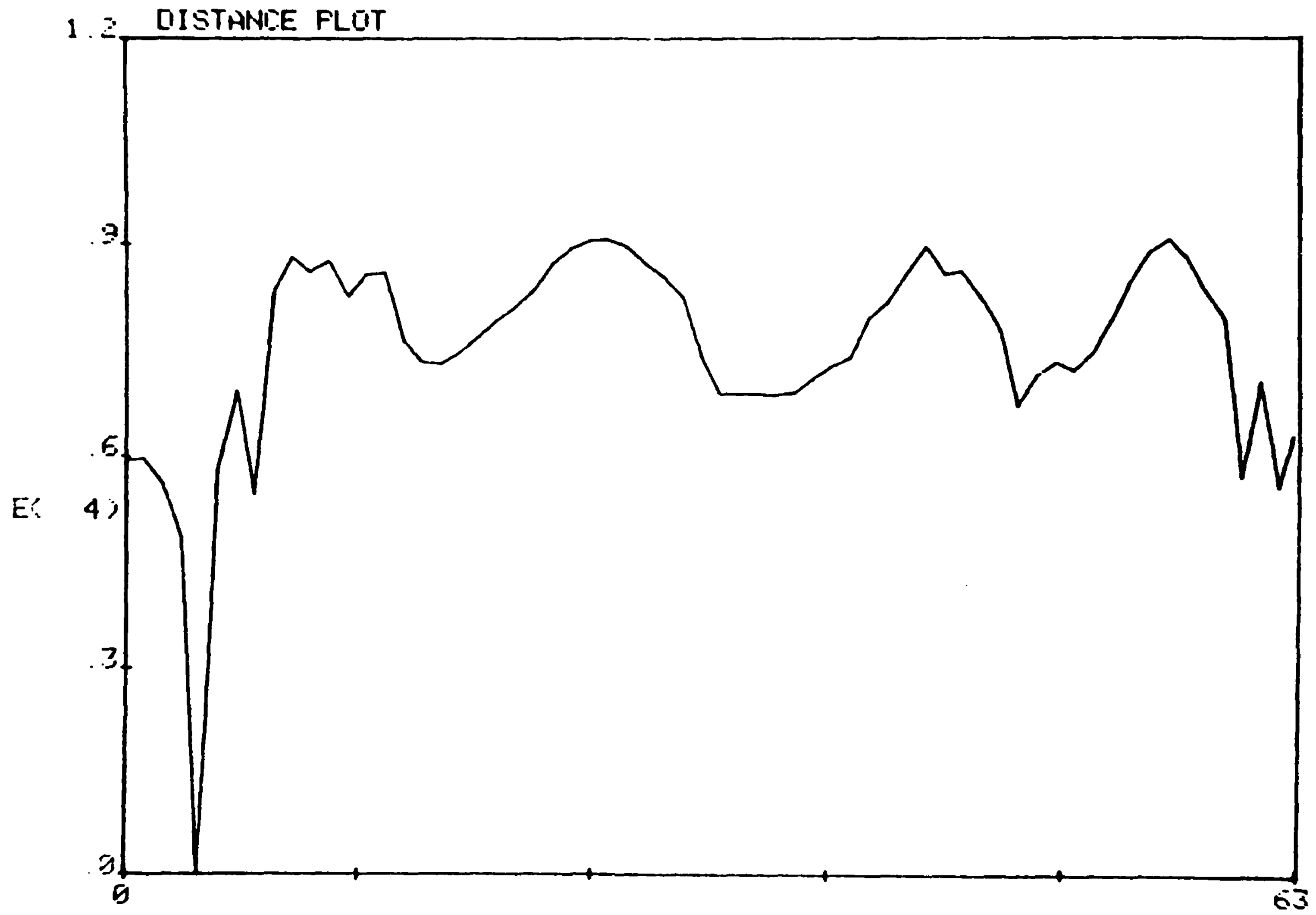


FIGURE 23. Distance plot of observation 85 against 81 through 144.
M2 rule, 64 point Hamming window, energy normalized.

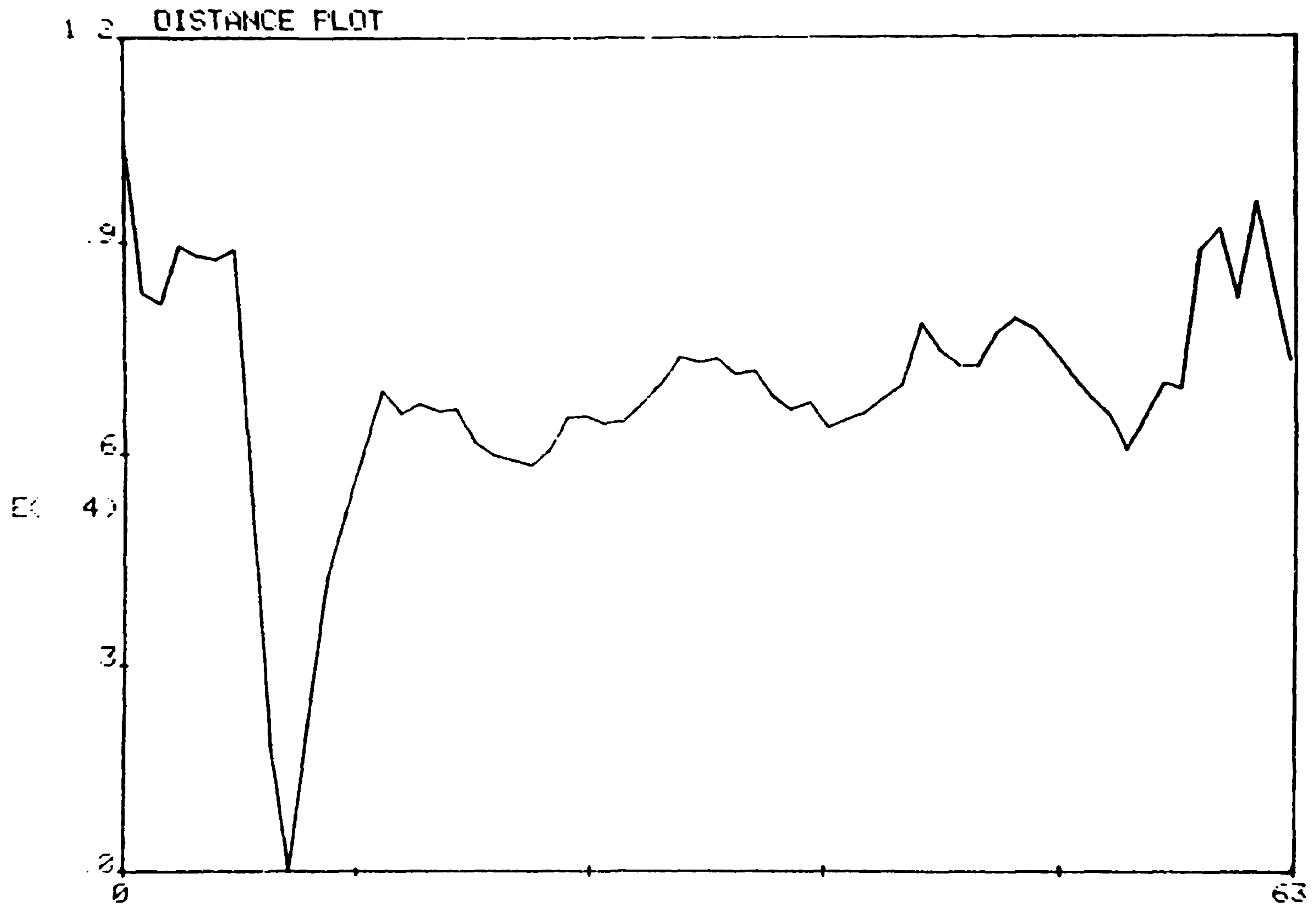


FIGURE 24. Distance plot of observation 90 against 81 through 144. M2 rule, 64 point Hamming window, energy normalized.

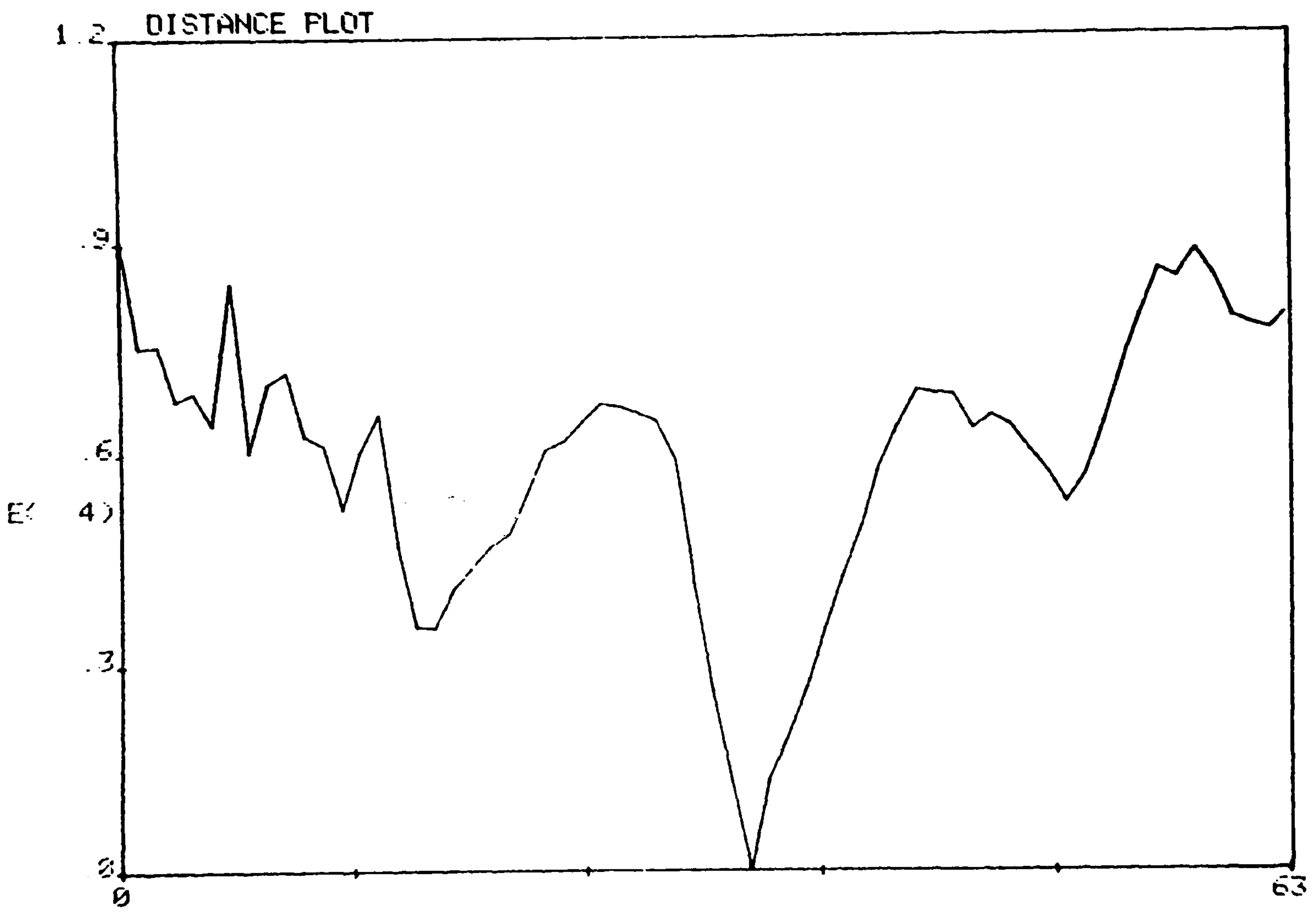


FIGURE 25. Distance plot of observation 115 against 81 through 144. M2 rule, 64 point Hamming window, energy normalized.

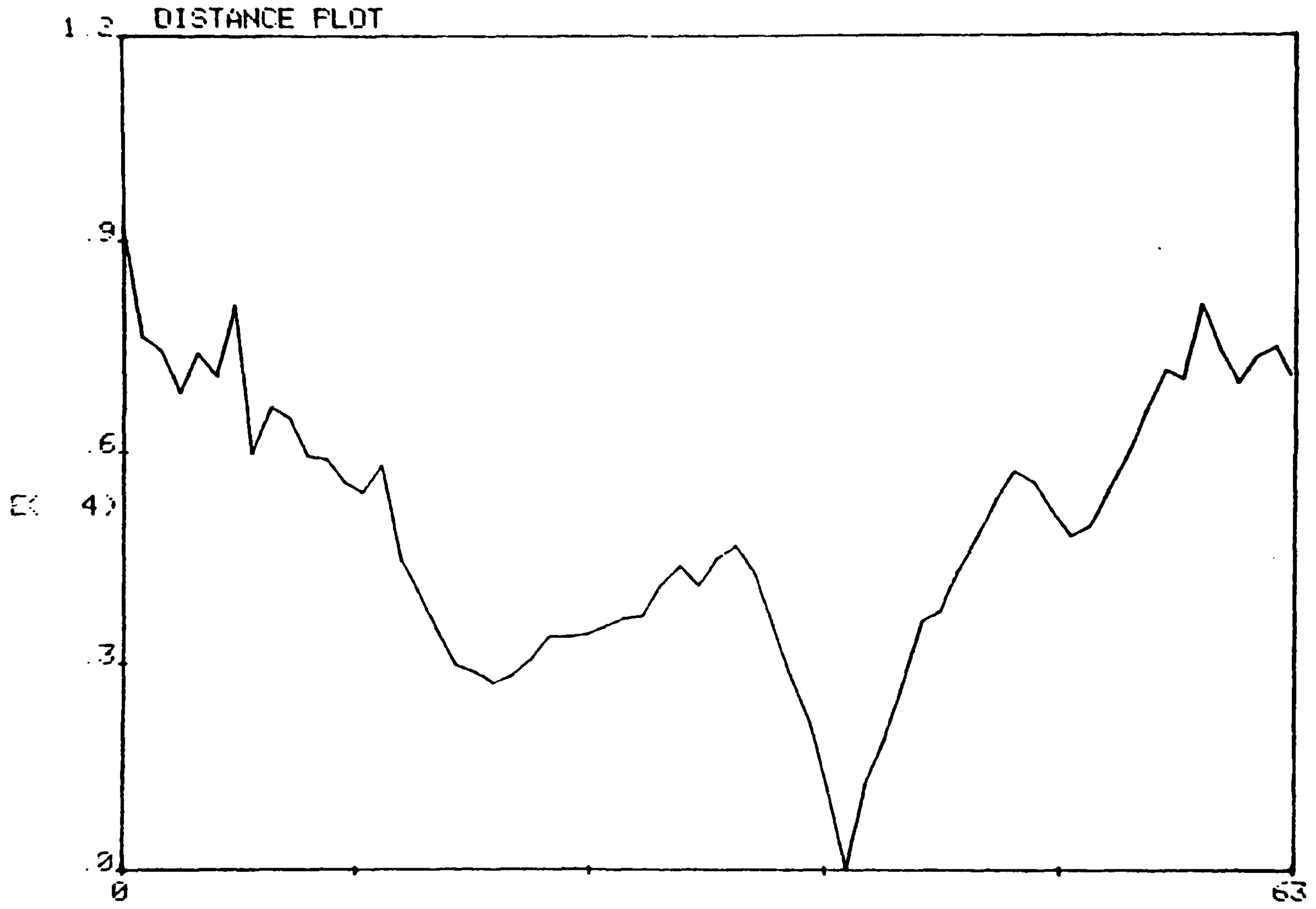


FIGURE 26. Distance plot of observation number 120 against 81 through 144. M2 rule, 64 point Hamming window, energy normalized.

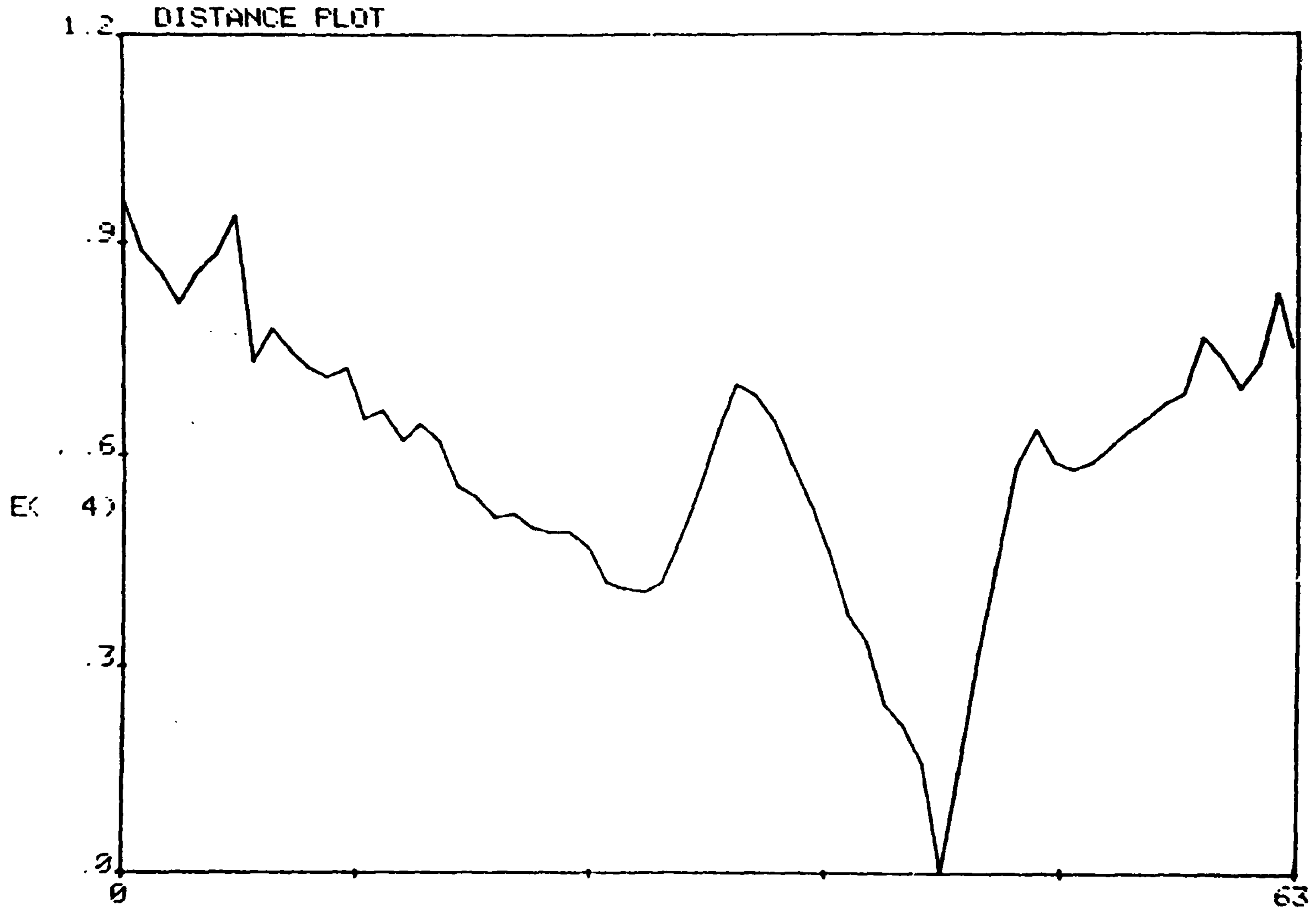


FIGURE 27. Distance plot of observation 125 against 81 through 144. M2 rule, 64 point Hamming window, energy normalized.

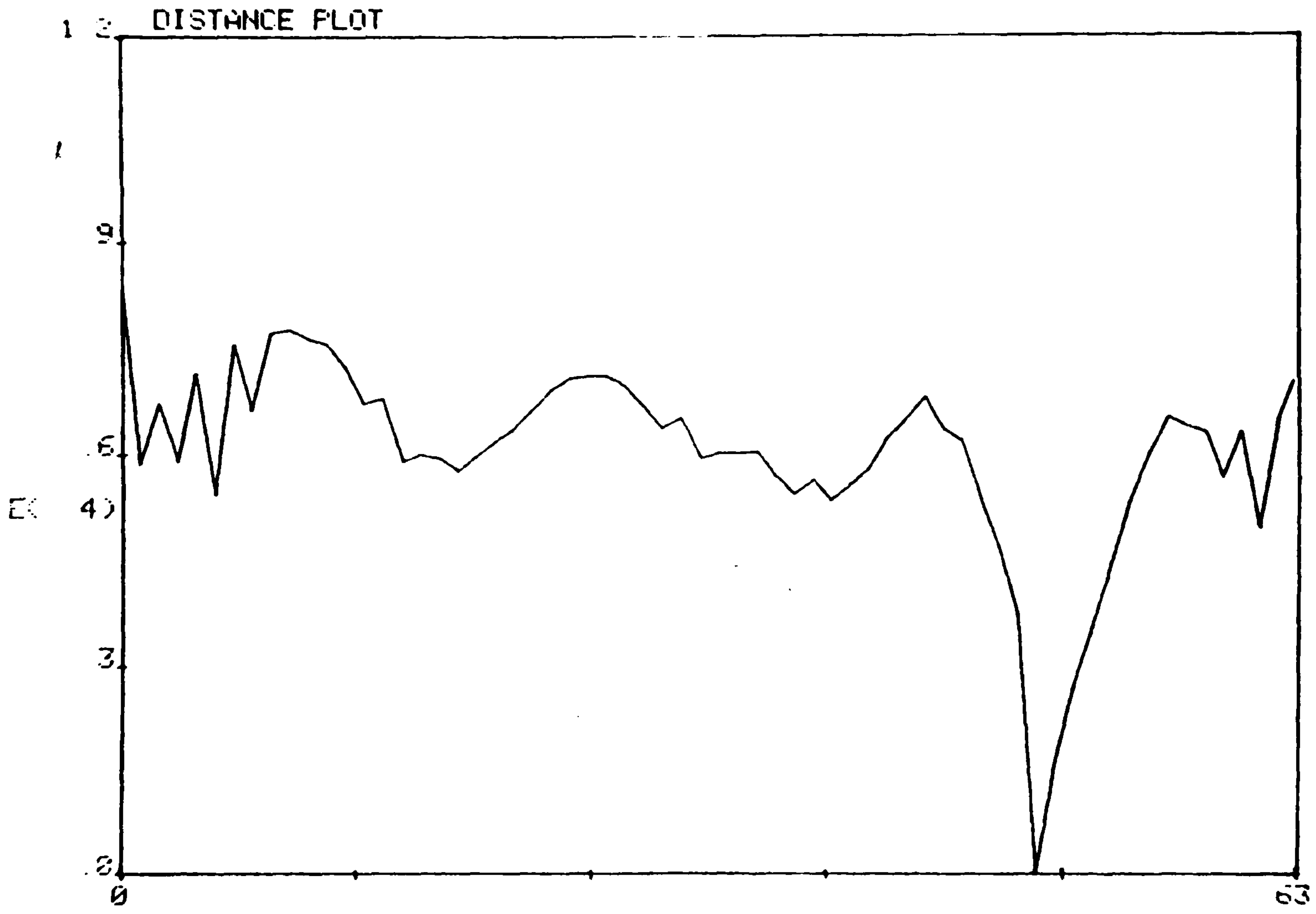


FIGURE 28. Distance plot of observation 130 against 81 through 144. M2 rule, 64 point Hamming window, energy normalized.

energy are emphasized. Sivian showed that formant amplitude rolled-off at approximately 10dB per Octave relative to the fundamental (Ref 11). Kuligowski has shown the same (Ref 12:20, 21). If pre-emphasis is applied to the speech spectrum to compensate for this roll-off, then sidelobes of the window spectrum excited by large amplitude fundamental and first formant components have a 20dB per Octave advantage over the high formants: 10dB per Octave because the formant is down that far and 10dB per Octave because the sidelobe is emphasized that much. De-emphasis of the fundamental in the spectral domain does not affect these sidelobes because they are generated in the transformation and de-emphasis is applied afterwards. They can be lowered by:

(1) High-pass filtering before the FFT to decrease the energy in the fundamental, and hence in the sidelobes.

(2) Using the longest possible window size and least distorting window shape while maintaining the 10MSEC-to-20MSEC window duration for short-time energy.

The first solution is difficult to apply. Kuligowski has shown (Ref 12:20, 21) that the location of the fundamental varies by as much as an octave over the vowels he listed, and depending on speaker sex and age. Further, for several vowel-speaker combinations, the fundamental of one vowel-speaker combination is at the same spectral location as the first formant at other vowel-speaker combinations. An adaptive filter before the FFT would be necessary to attenuate the fundamental prior to transformation without attenuating the first formant. The second solution can be implemented with enough memory and a fast enough analog-to-digital converter. At a 1024 point window size, a sampling rate of 68KHz would provide a window duration of 15MSEC.

As has been previously noted, the FFT is a robust feature (Ref 7). It may be sufficiently robust for an 8KHz sampling rate and a 128 point Hamming window with the observation pre-emphasized at 10dB per Octave. The measure of sufficiency is in the way the features cluster, and an adequate feature-space partitioning algorithm for this problem has not yet been implemented.

It is clear that the fundamental should be de-emphasized in the observation spectrum. It is a high energy feature which is highly speaker dependent. The problem with de-emphasizing it is that for some vowel-speaker combinations, it is located at or near the first formant of other vowel-speaker combinations (Ref 12:20, 21). We chose to de-emphasize at 10dB/Octave ending at a corner frequency of 300Hz as a compromise between allowing too much fundamental energy to remain in the observation, and reducing the amount of energy in the important first formant.

THRESHOLDING:

The last parameter Seelandt identified for further study is a threshold below which phonetic units should be attenuated when their energy content is low. When the energy in a time slice did not exceed a preset threshold, the spectral components were attenuated rather than energy normalized. Seelandt did this to prevent vectors which consist predominately of background noise from entering into the decision process (Ref 1:121, 122).

We found that unvoiced fricative sounds, especially those in sounds involving the letter "f," are low energy sounds and hard to distinguish from noise in a spectrogram. Phonet plots did reveal structure in these sounds that might distinguish them from white noise (flat spectrum), or other background noise models. Distance measures computed to phonets which represent background noise, coupled with knowledge of phonet and observation energy, may permit a phonet-to-word translation machine, such as Montgomery's (Ref 10), to distinguish between unvoiced fricative sounds and noise, and may aid in detecting word boundaries as well. Figures 29-55 illustrate observations taken from regions of the speech file CT56.SP known to contain noise as well as the words "five" and "six." The plots are in dB versus spectral component number. Component number zero is the observation energy, also in dB. Figures 29-34 are from a region where there was no speaker sound. Compare those figures with Figures 35-42 which are from a region of the file where the speaker begins the "five" sound. It is difficult to detect where the "f" sound begins, either by studying the spectral distribution or by noting the energy. One can see that as the characteristic formant structure associated with the vowel "i" sound develops, the energy increases from roughly 63dB to roughly 100dB in the "i" sound. But it does not appear that energy content is a useful means for discriminating between noise and the "f" sound.

Figures 43-50 were taken from the word "five" from well into the vowel sound "i" through the "v" sound on into a pause between the words "five" and "six." One can see the spectral distribution change from the "i" formant structure through a "v" sound structure of relatively flat spectrum but moderate energy, to a nonspeaker background noise of flat spectrum and low energy. Figures 51-56 show a sequence of observations from the speakerless background region between the words to well into the "s" in "six." Note that as the characteristic upward slope (left-to-right) of the spectral distribution of the "s" sound develops, the energy increases roughly 25dB. One can see in this overall sequence of figures that energy is a poor discriminant between noise and some unvoiced sounds. We conclude that energy is an unreliable discriminant between background noise and unvoiced speech.

The question of how unvoiced sounds may be detected in noise needs to be addressed. It is beyond the scope of this project to do so because it remains to be seen if

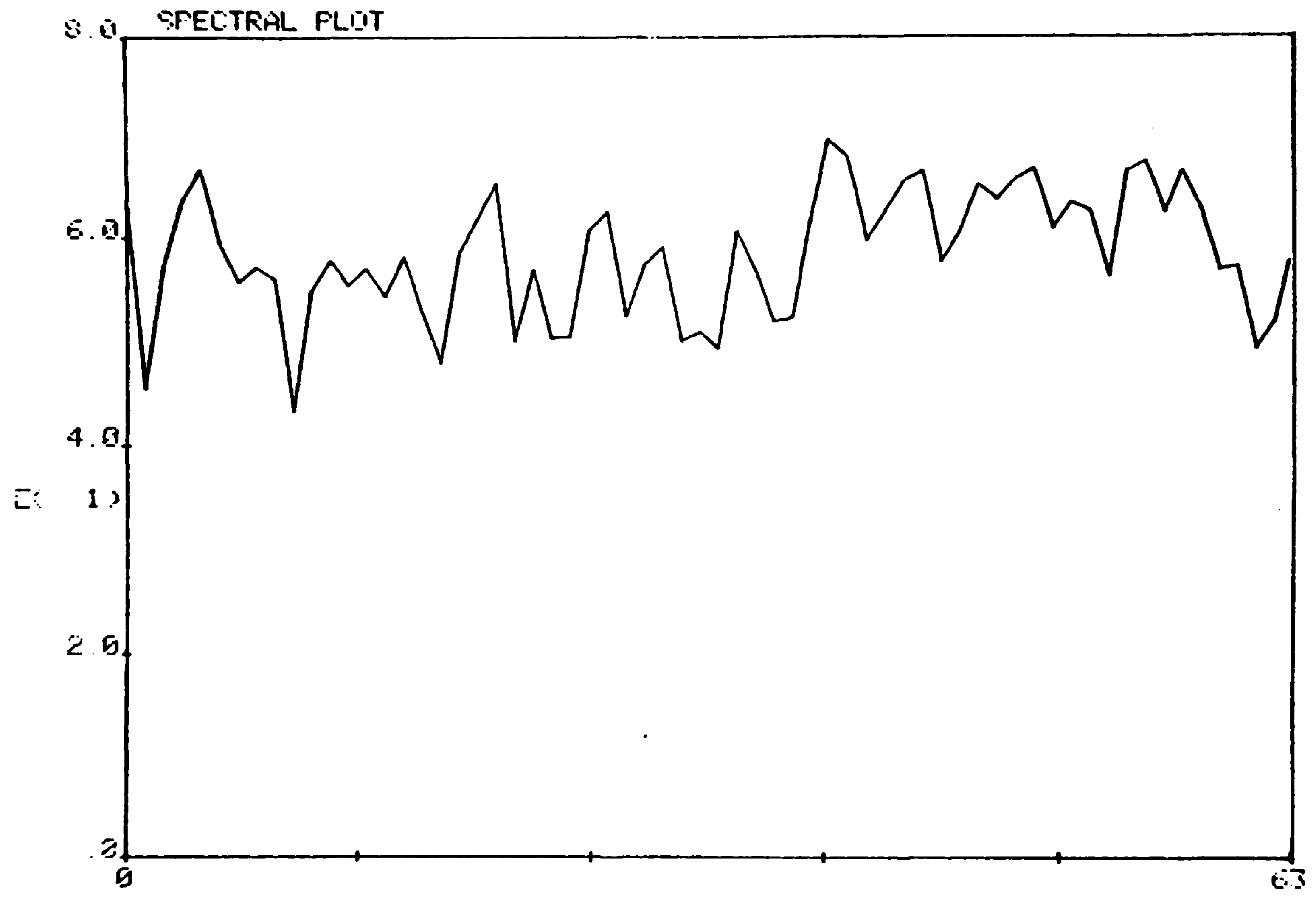


FIGURE 29. Spectral plot observation number 21, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

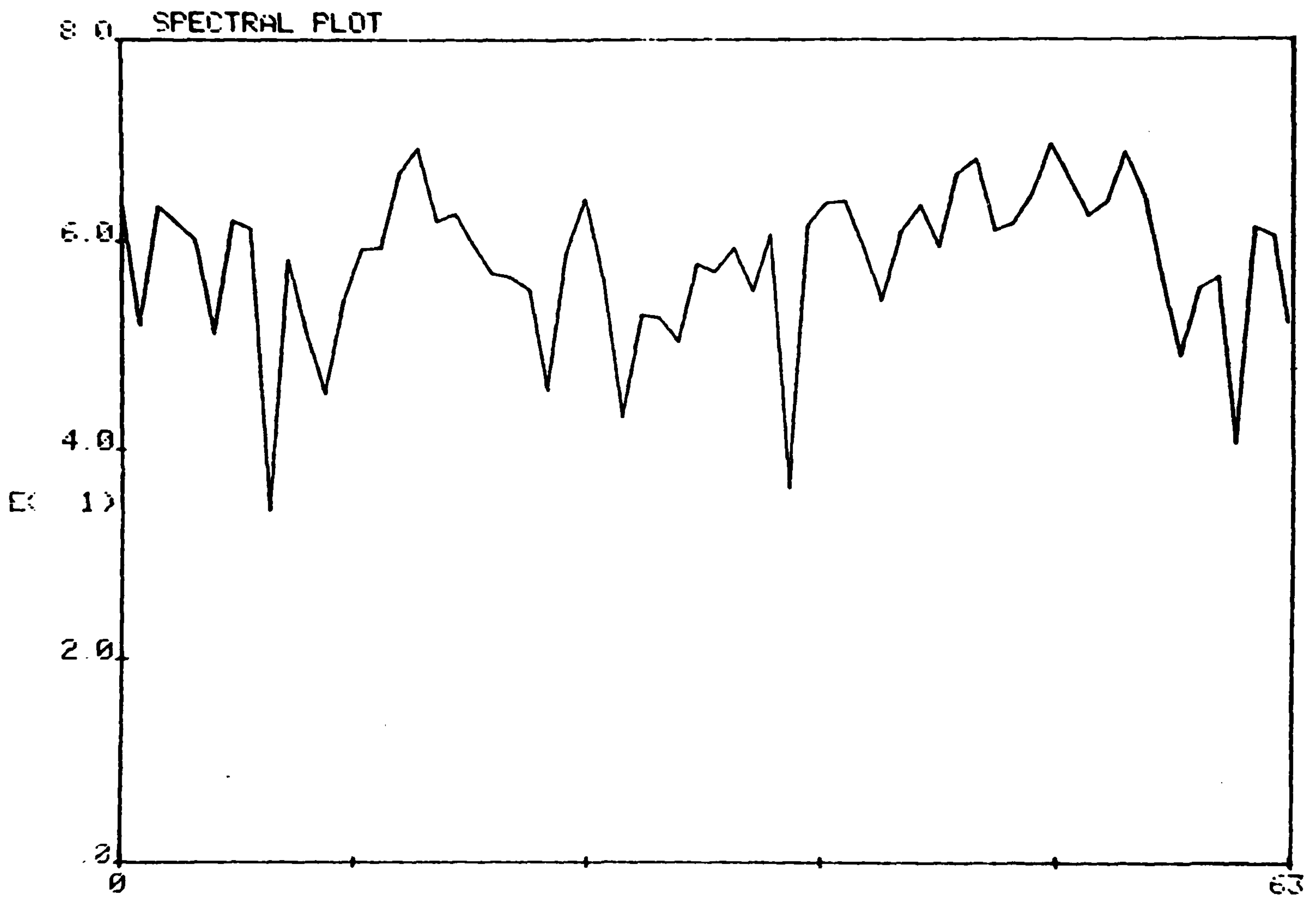


FIGURE 30. Spectral plot observation number 24, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

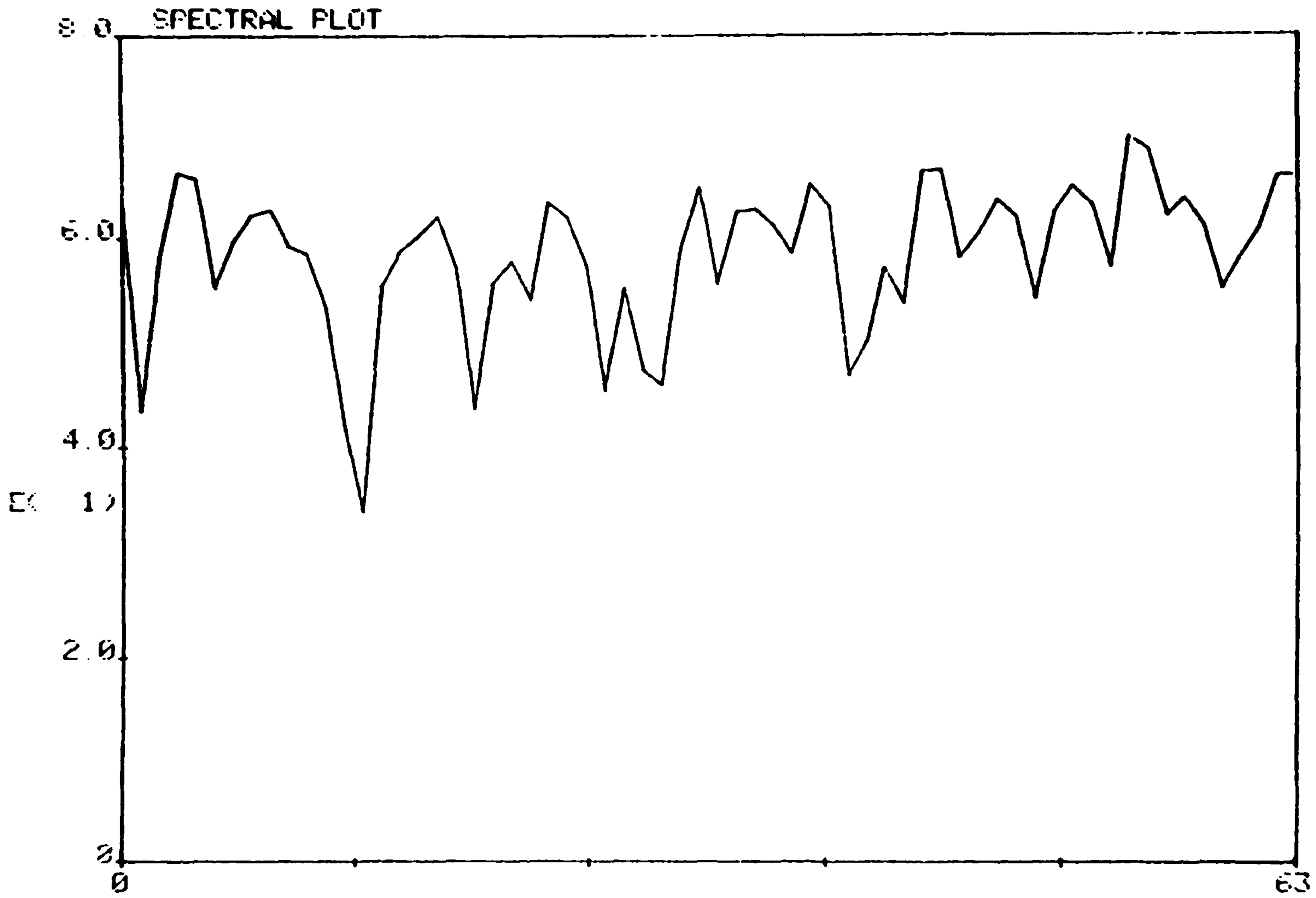


FIGURE 31. Spectral plot observation number 31, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

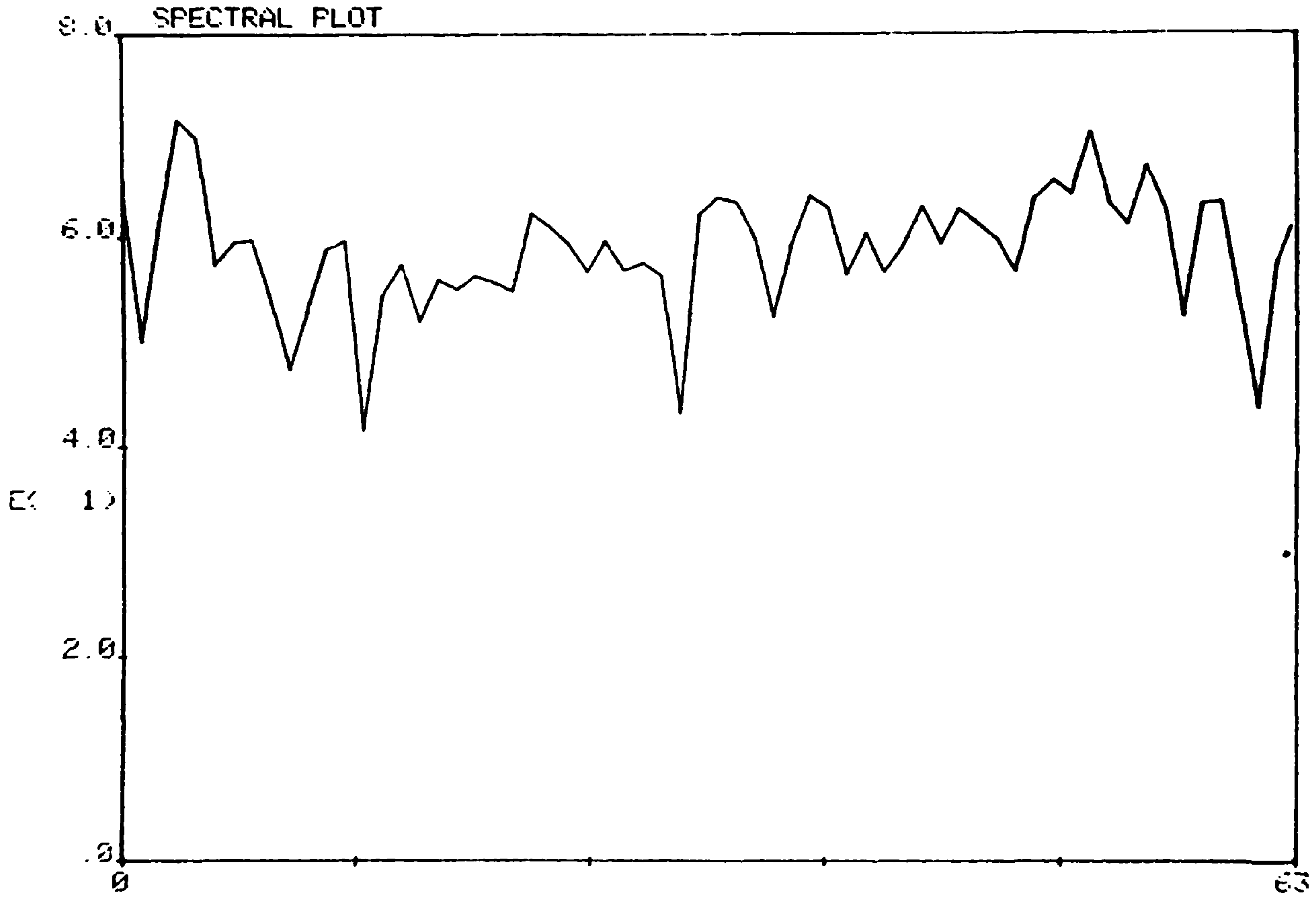


FIGURE 32. Spectral plot observation number 34, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

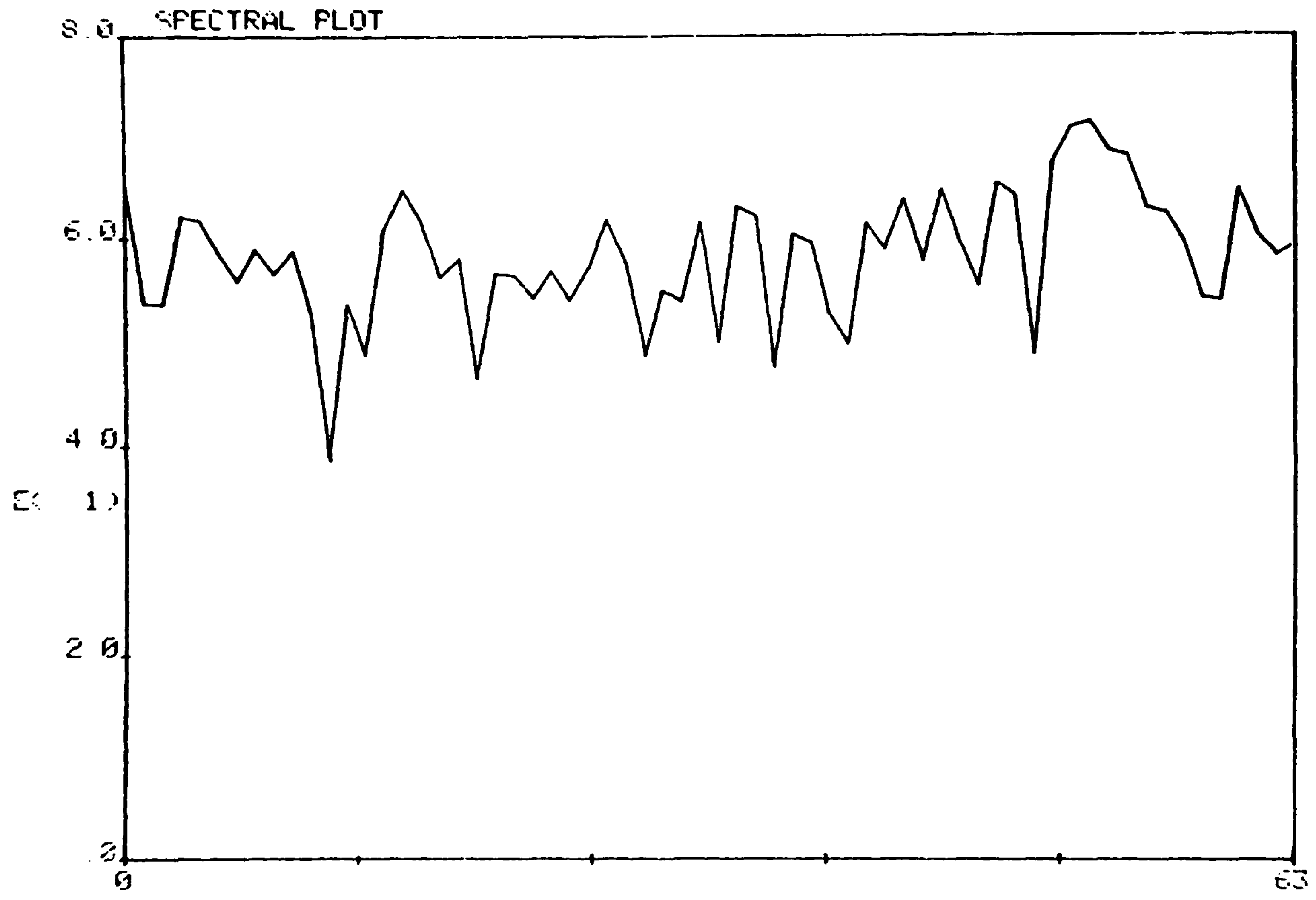


FIGURE 33. Spectral plot observation number 70, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

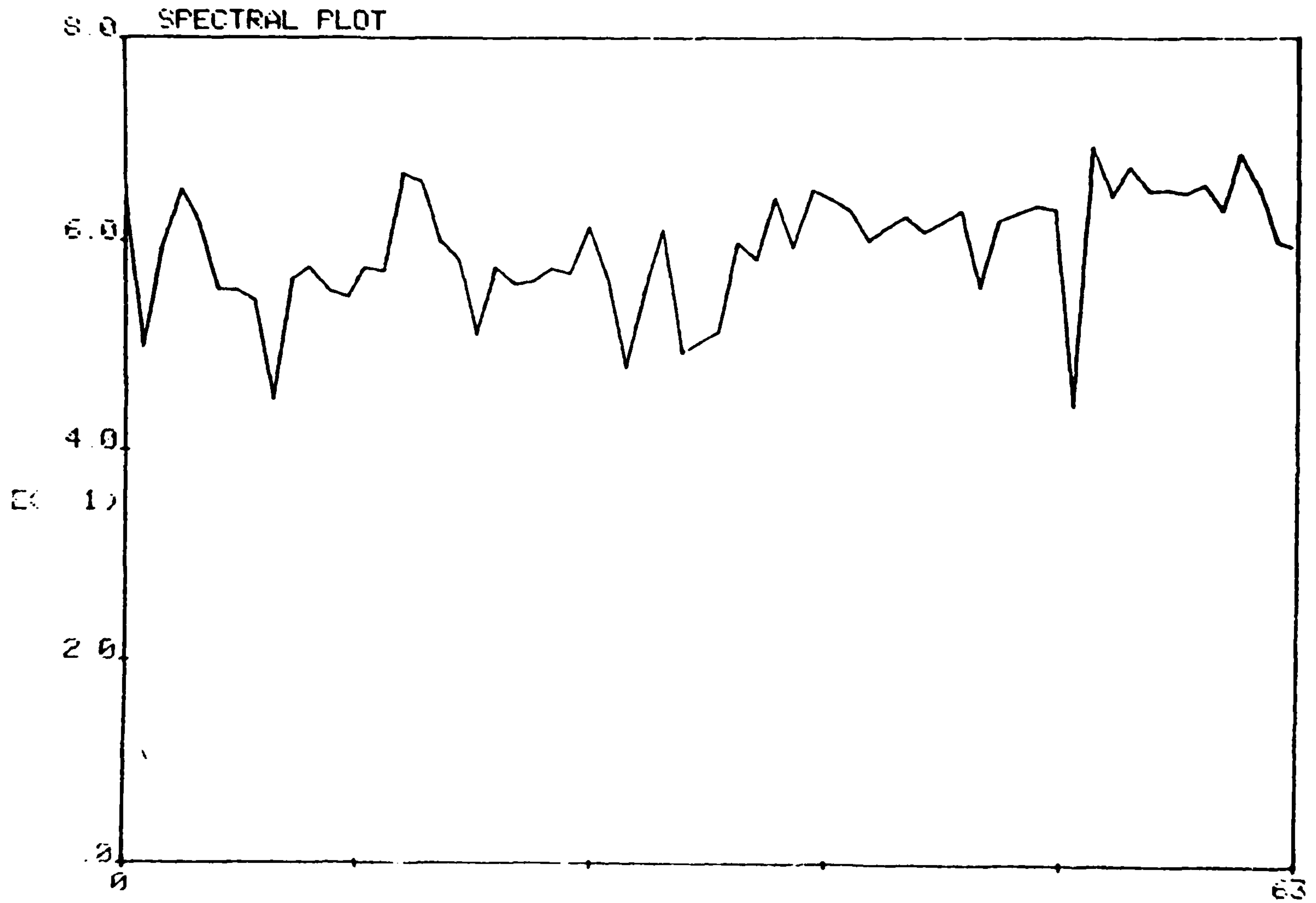


FIGURE 34. Spectral plot observation number 75, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

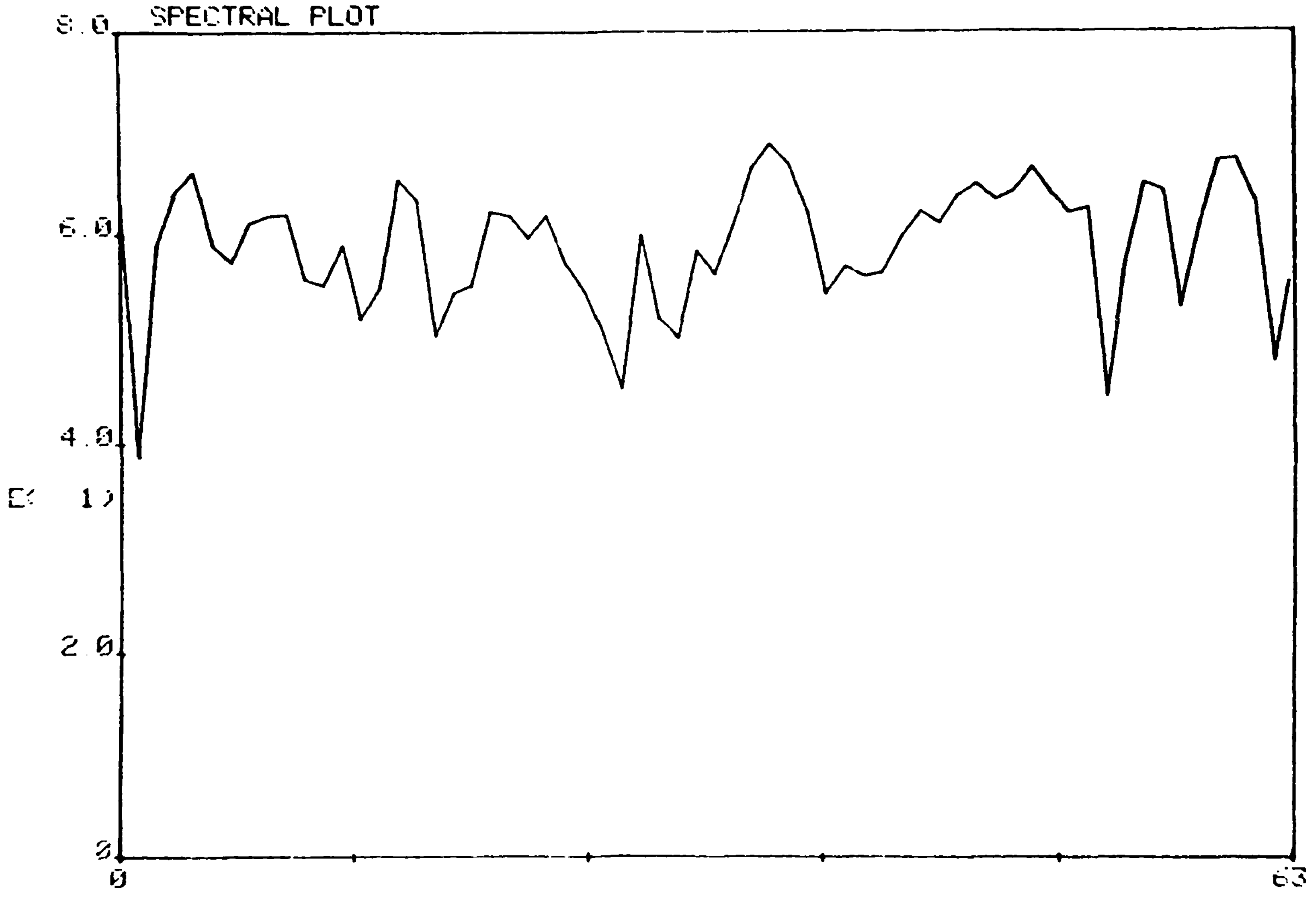


FIGURE 35. Spectral plot observation number 80, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

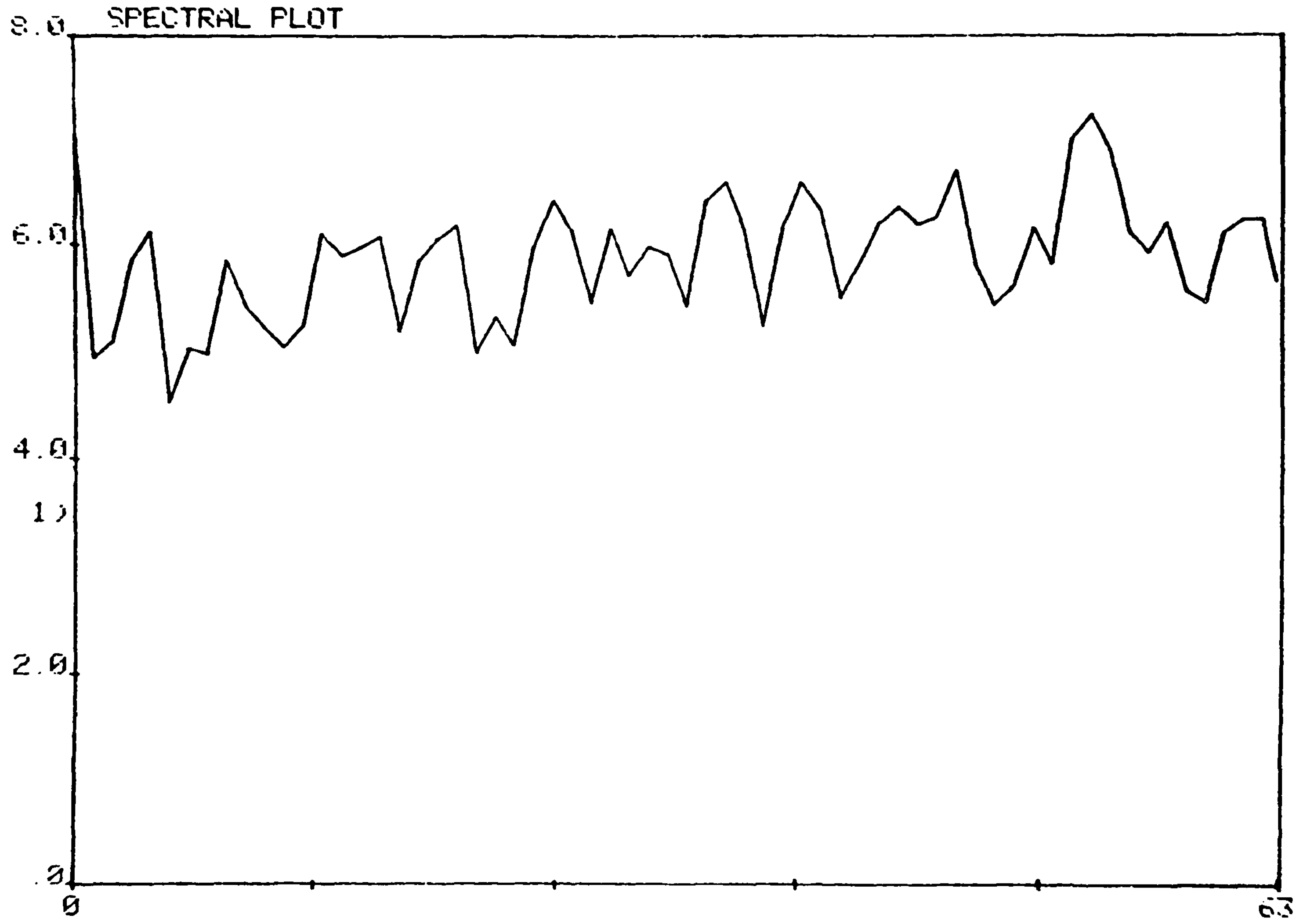


FIGURE 36. Spectral plot observation number 85, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

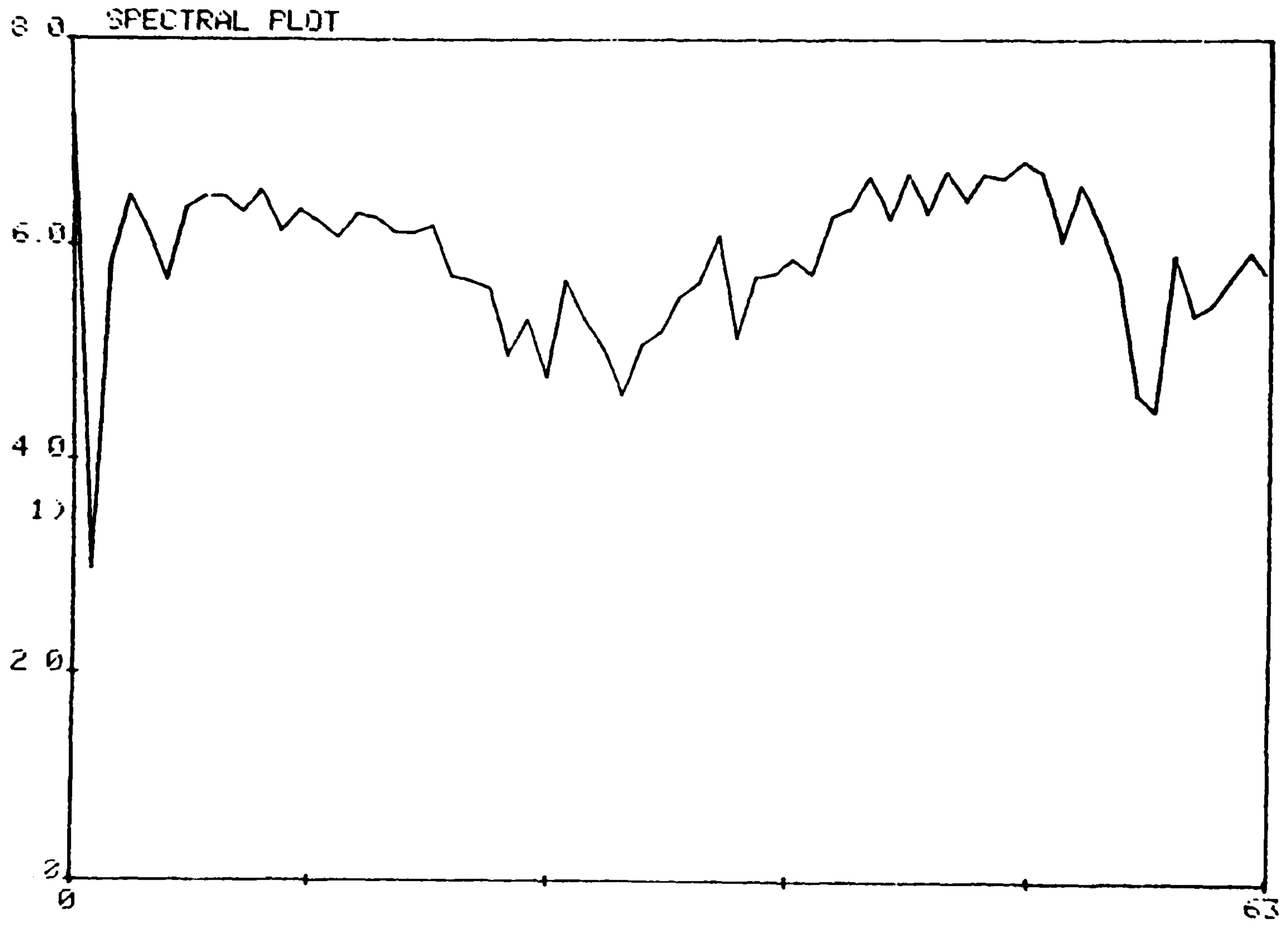


FIGURE 37. Spectral plot observation number 87, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

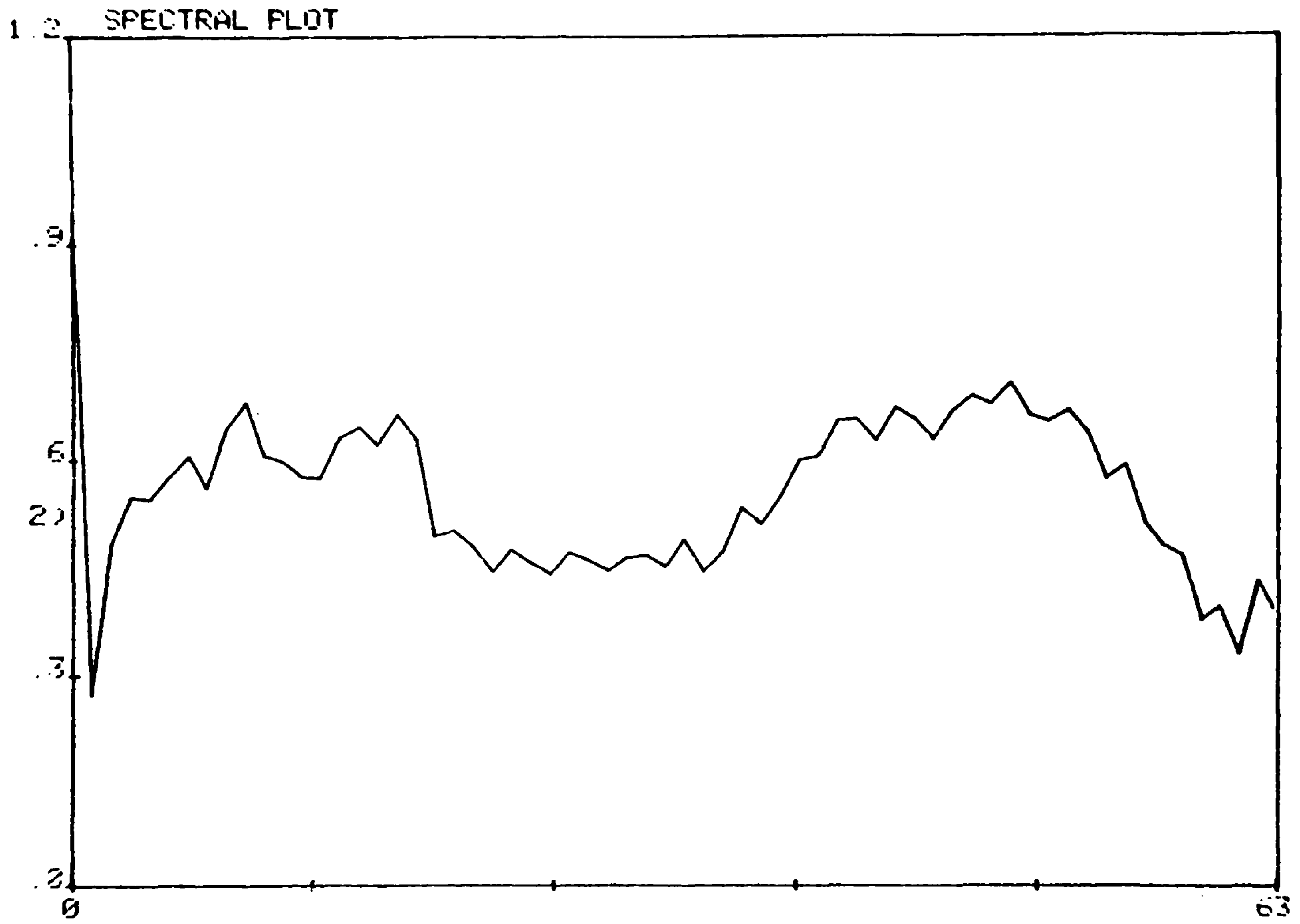


FIGURE 38. Spectral plot observation number 88, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

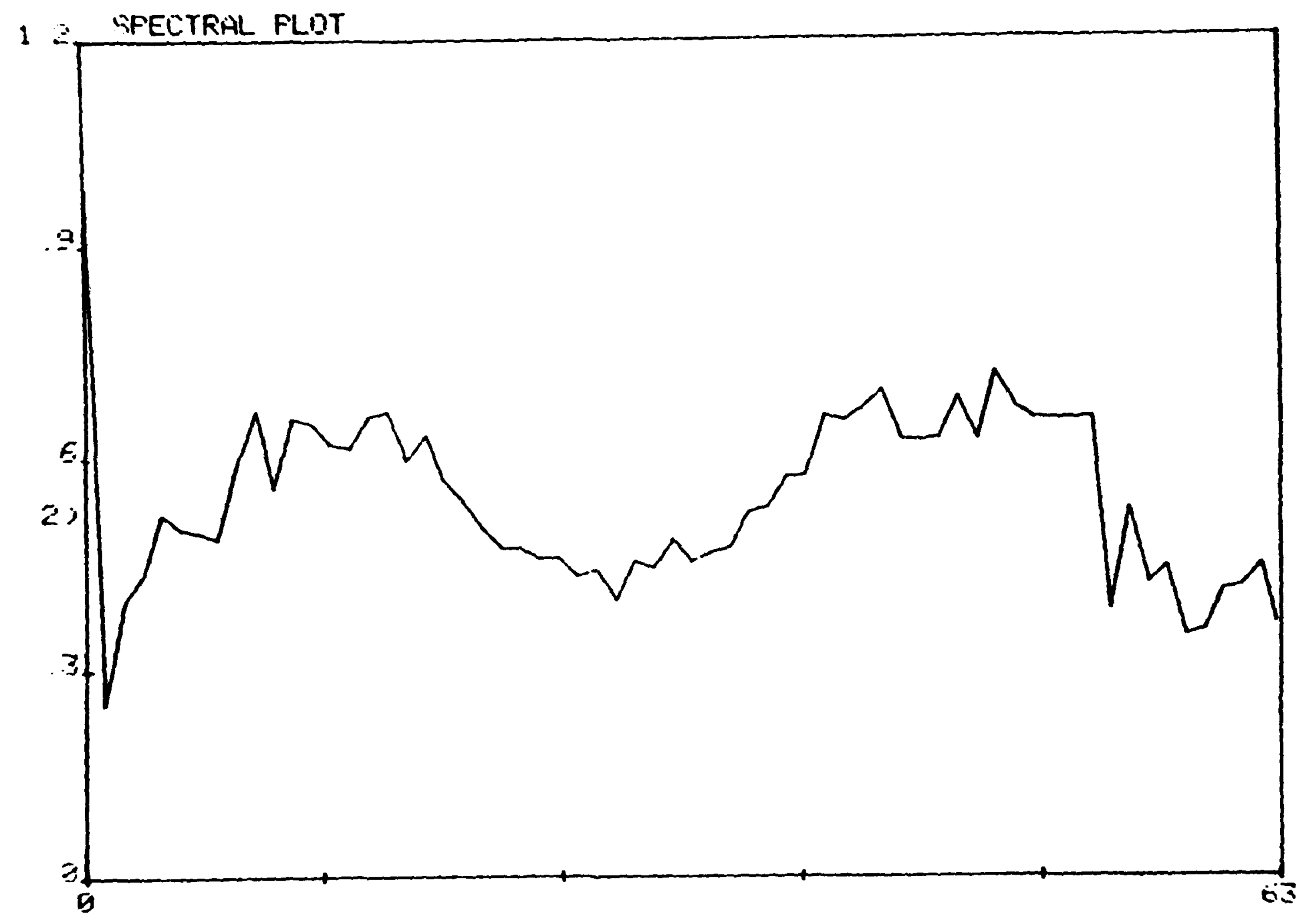


FIGURE 39. Spectral plot observation number 90, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

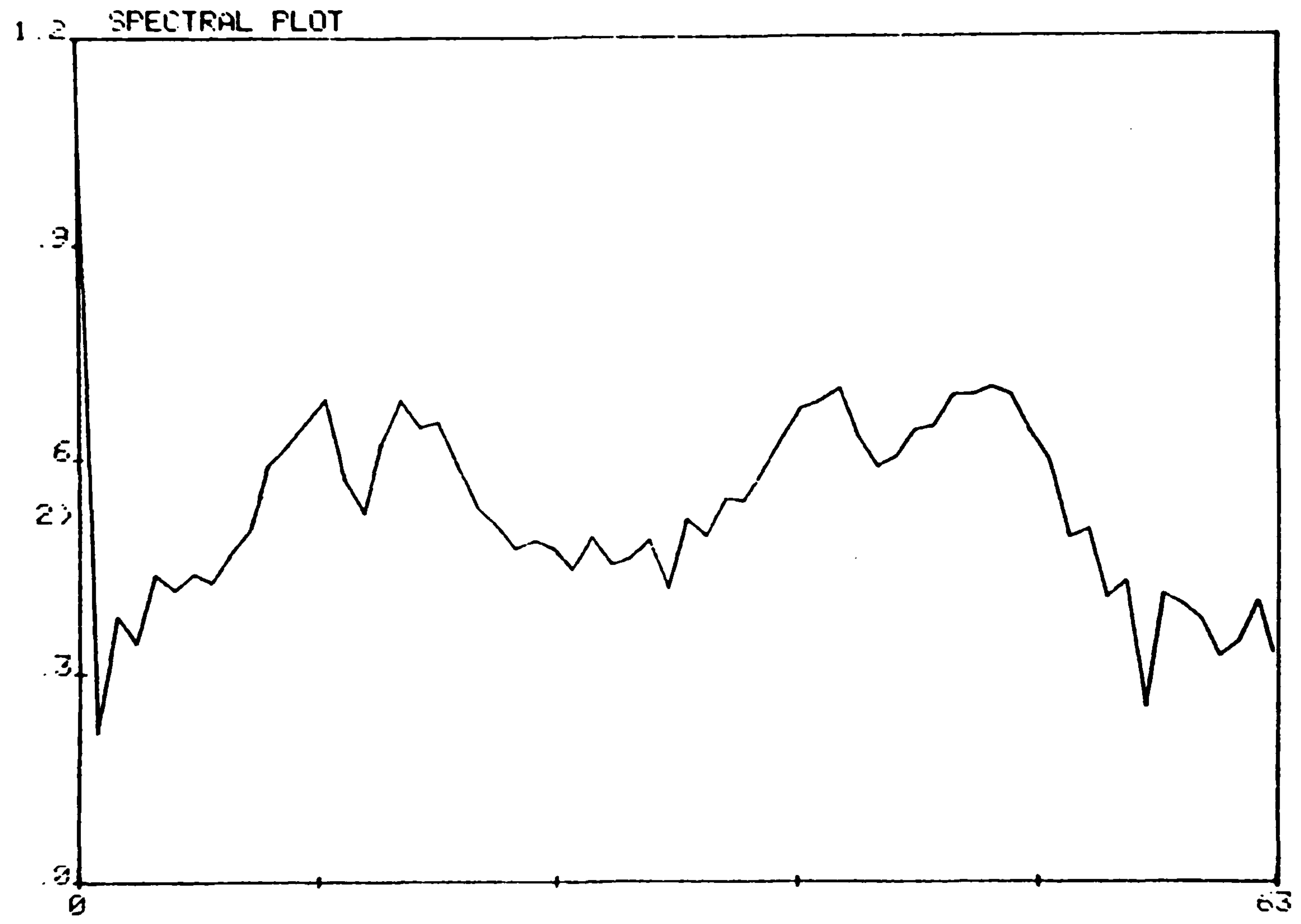


FIGURE 40. Spectral plot observation number 95, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

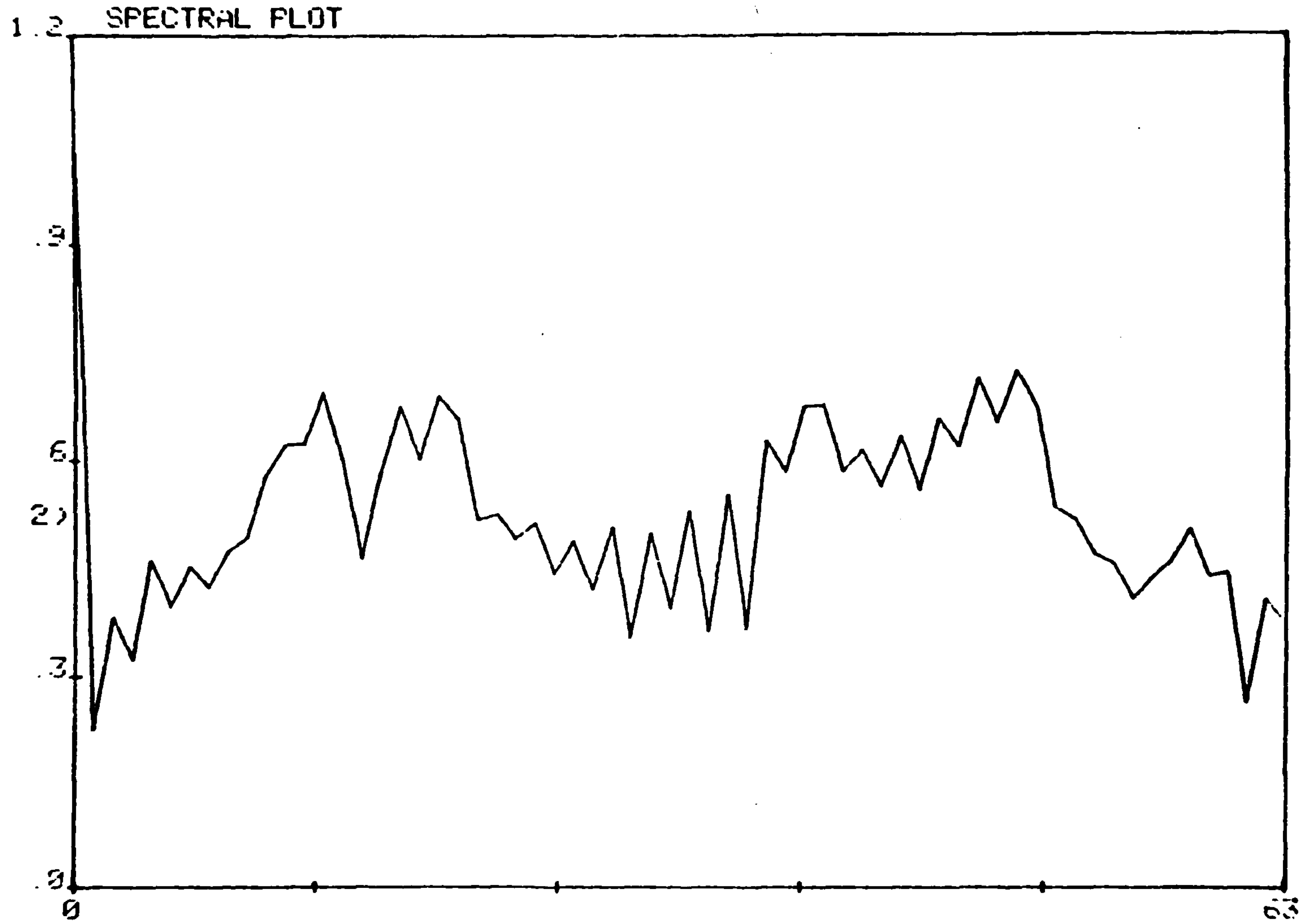


FIGURE 41. Spectral plot observation number 100, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

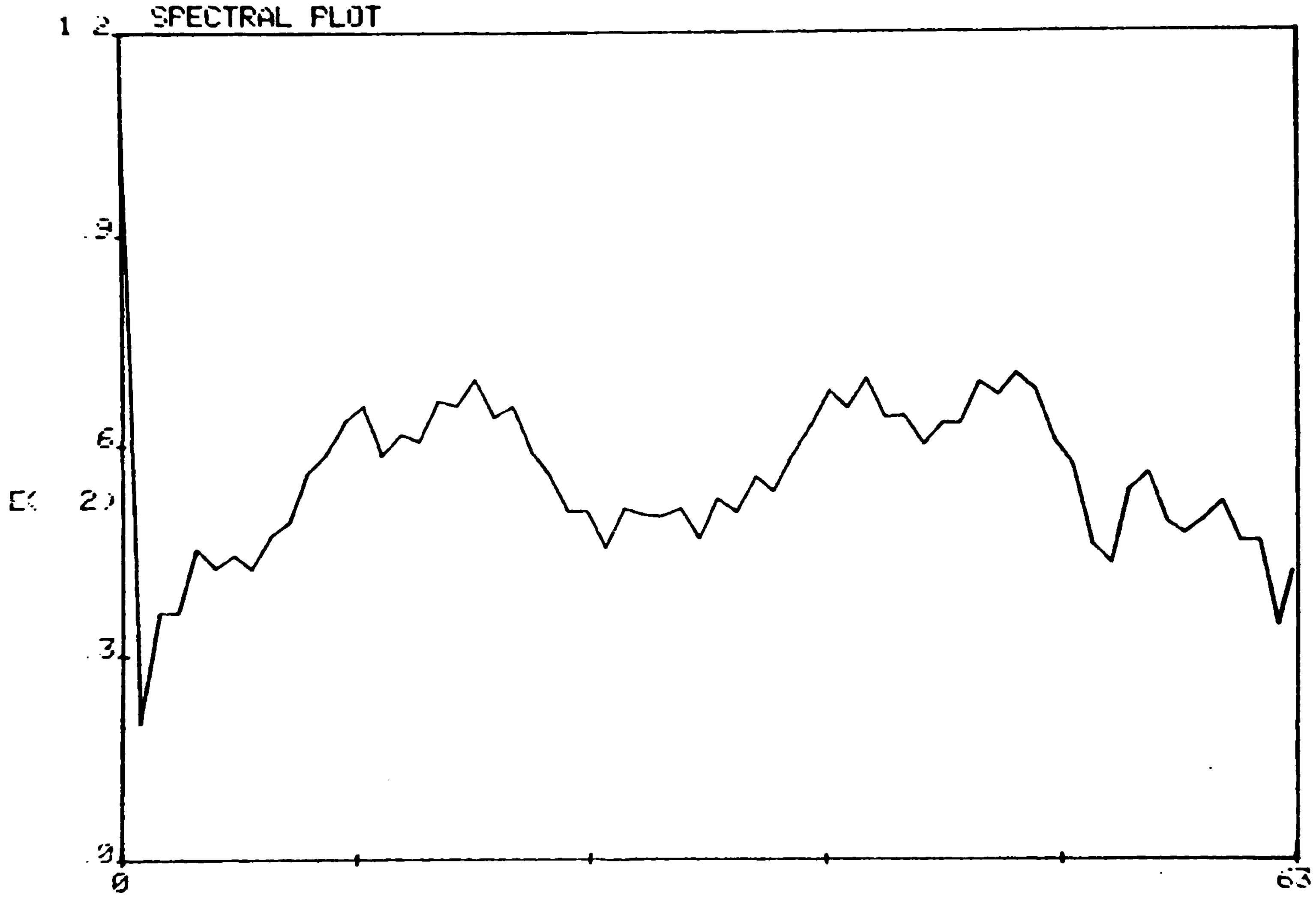


FIGURE 42. Spectral plot observation number 110, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

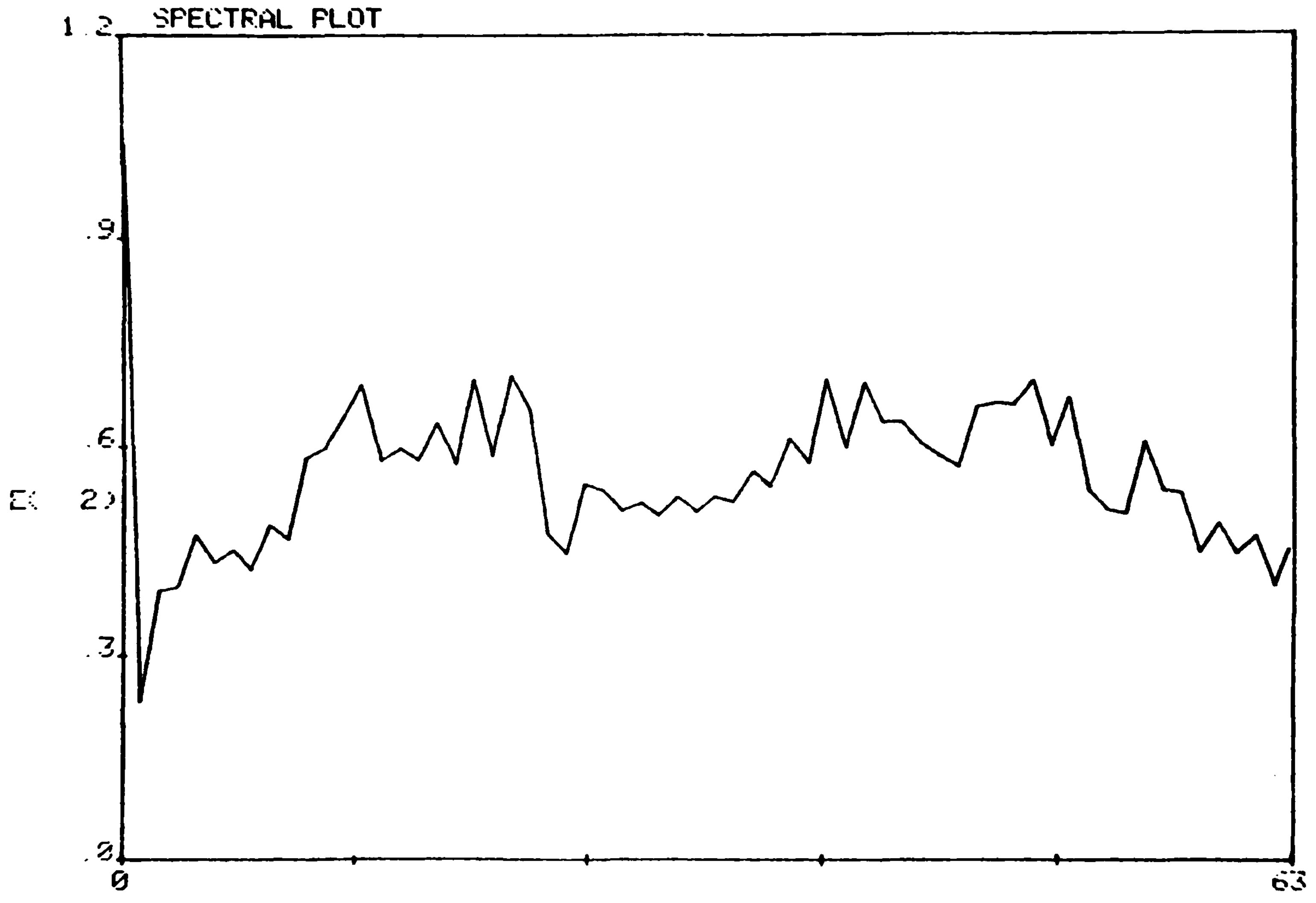


FIGURE 43. Spectral plot observation number 120, 128 point overlapping Hamming window, 10dB de-emphasis to 300FZ, 10dB pre-emphasis from 500HZ, energy normalized.

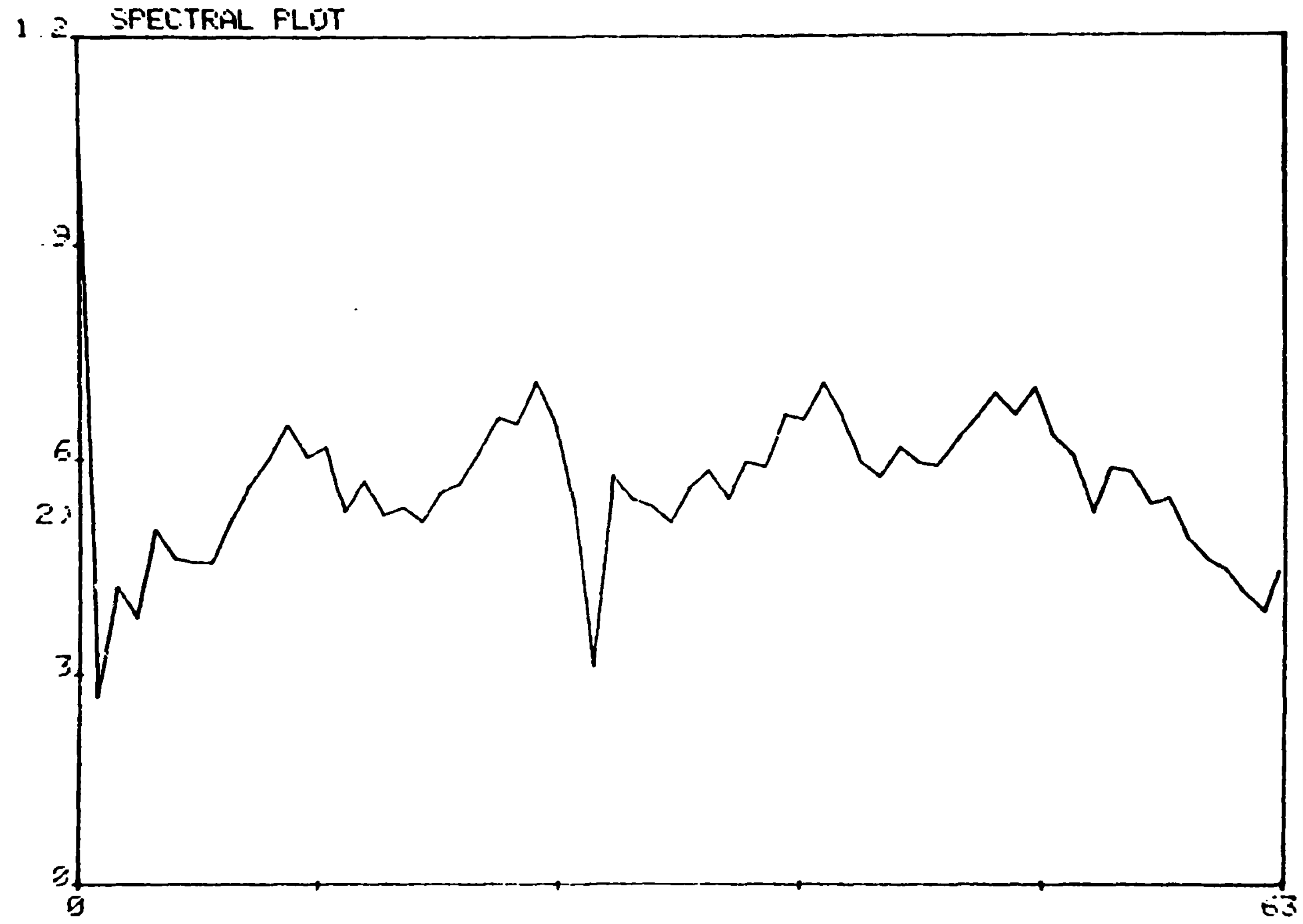


FIGURE 44. Spectral plot observation number 130, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

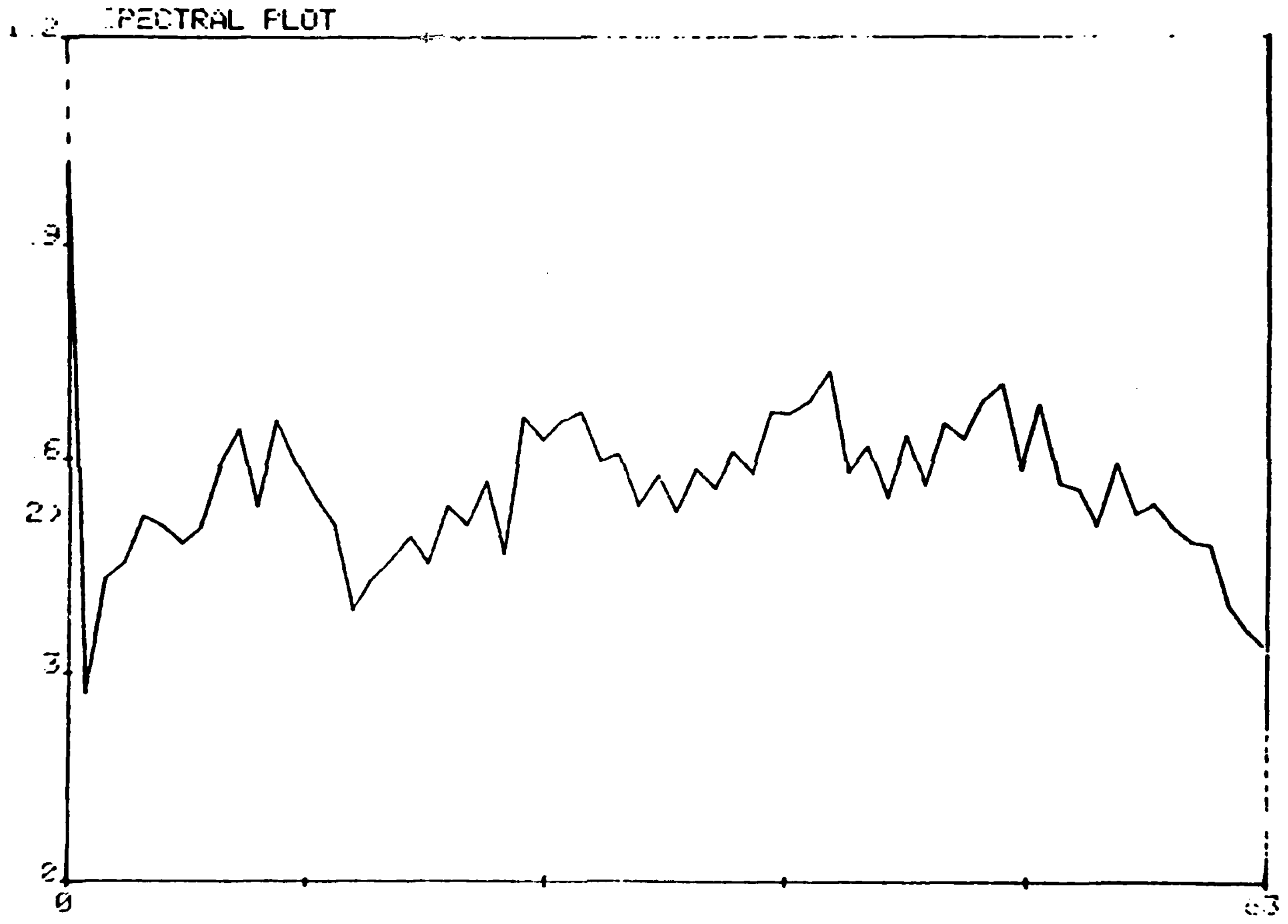


FIGURE 45. Spectral plot observation number 135, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

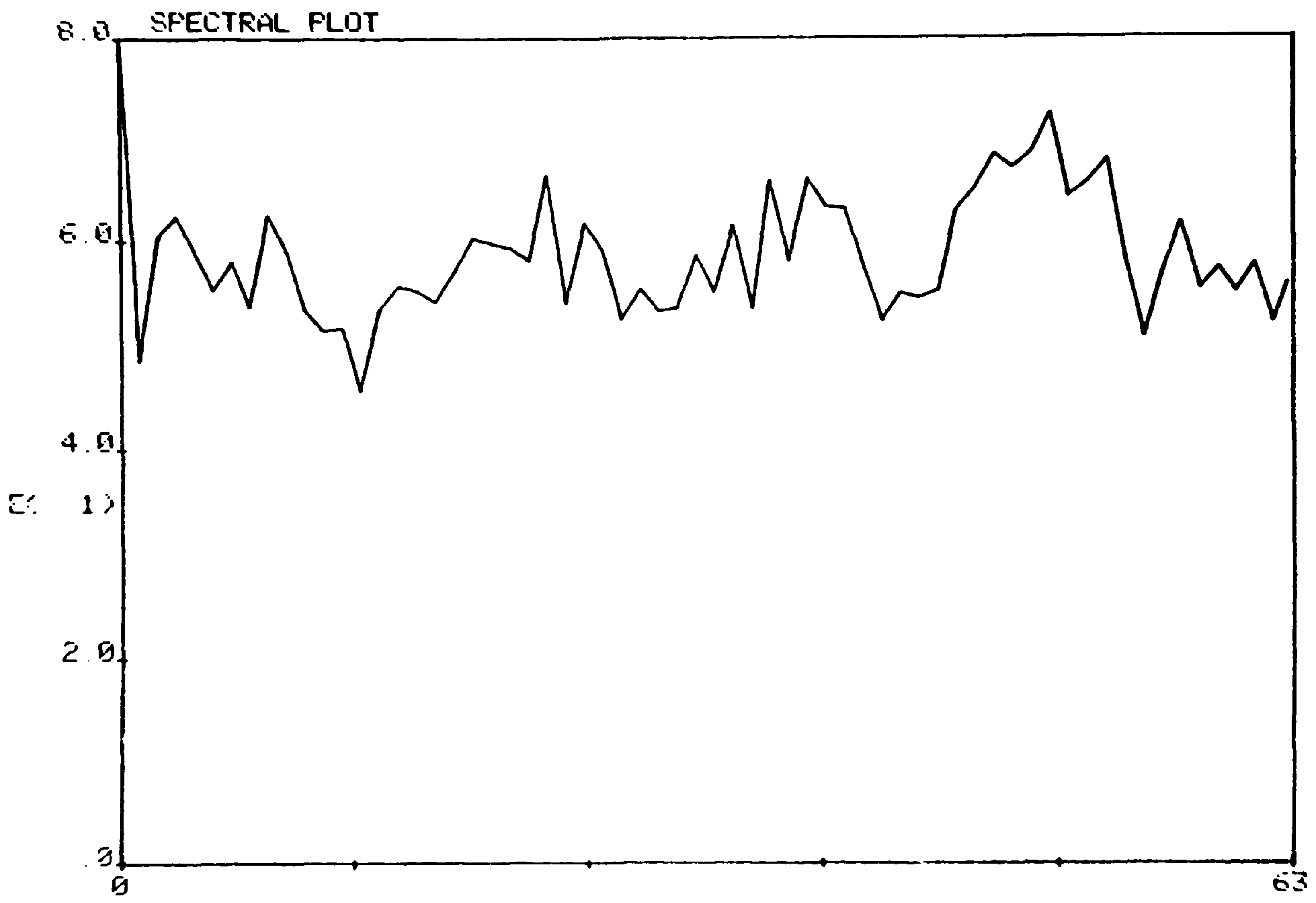


FIGURE 46. Spectral plot observation number 140, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

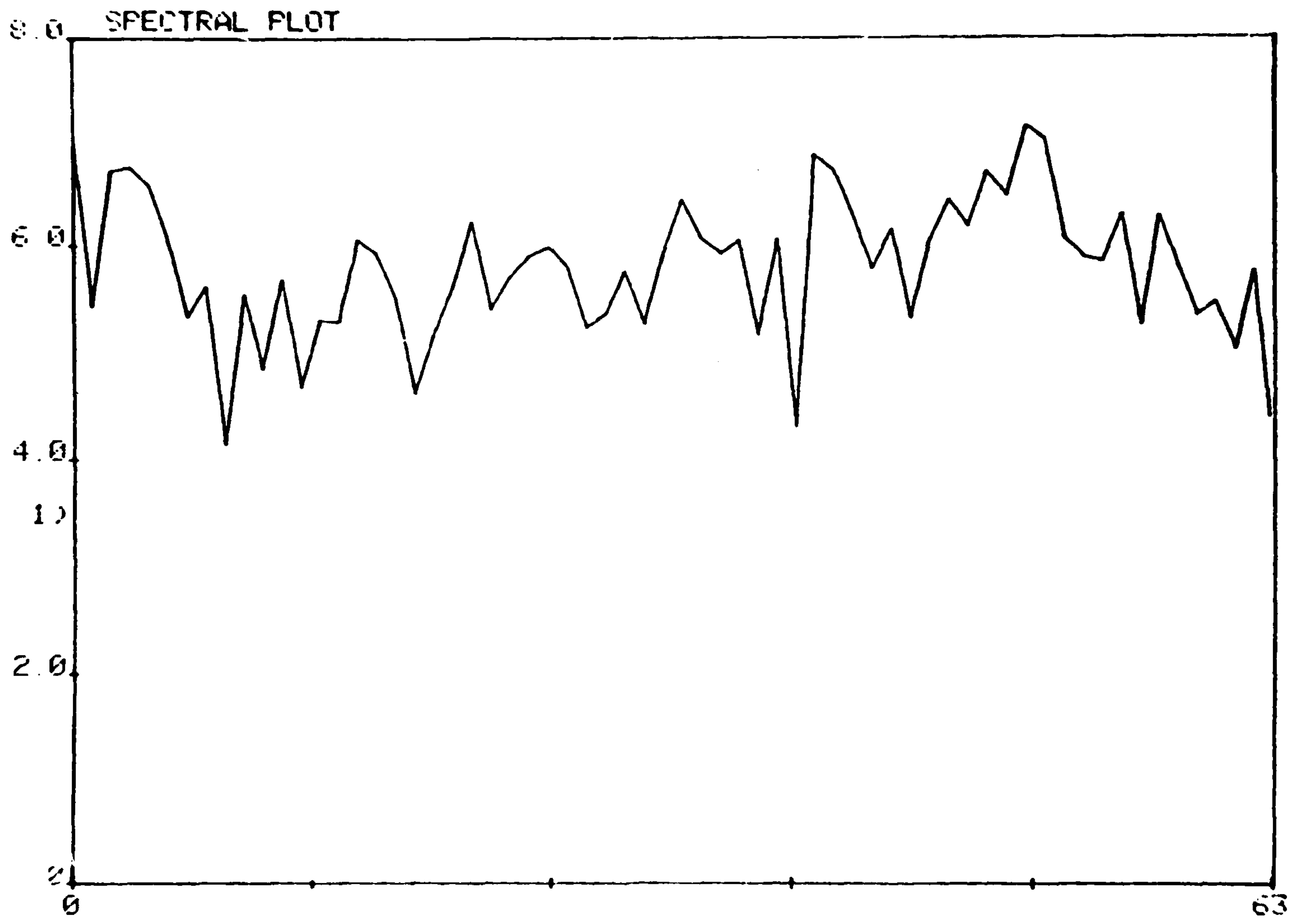


FIGURE 47. Spectral plot observation number 145, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

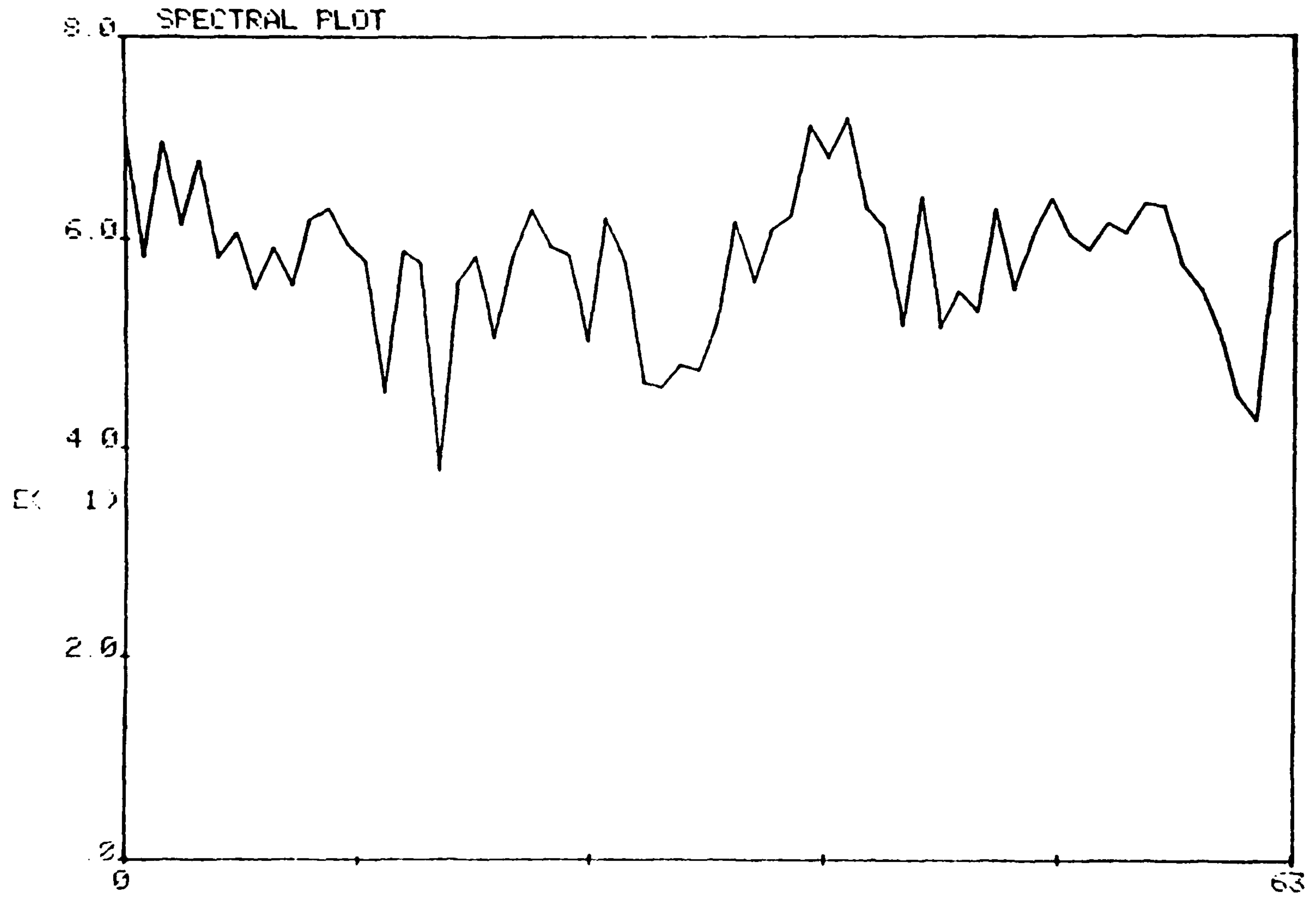


FIGURE 48. Spectral plot observation number 150, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

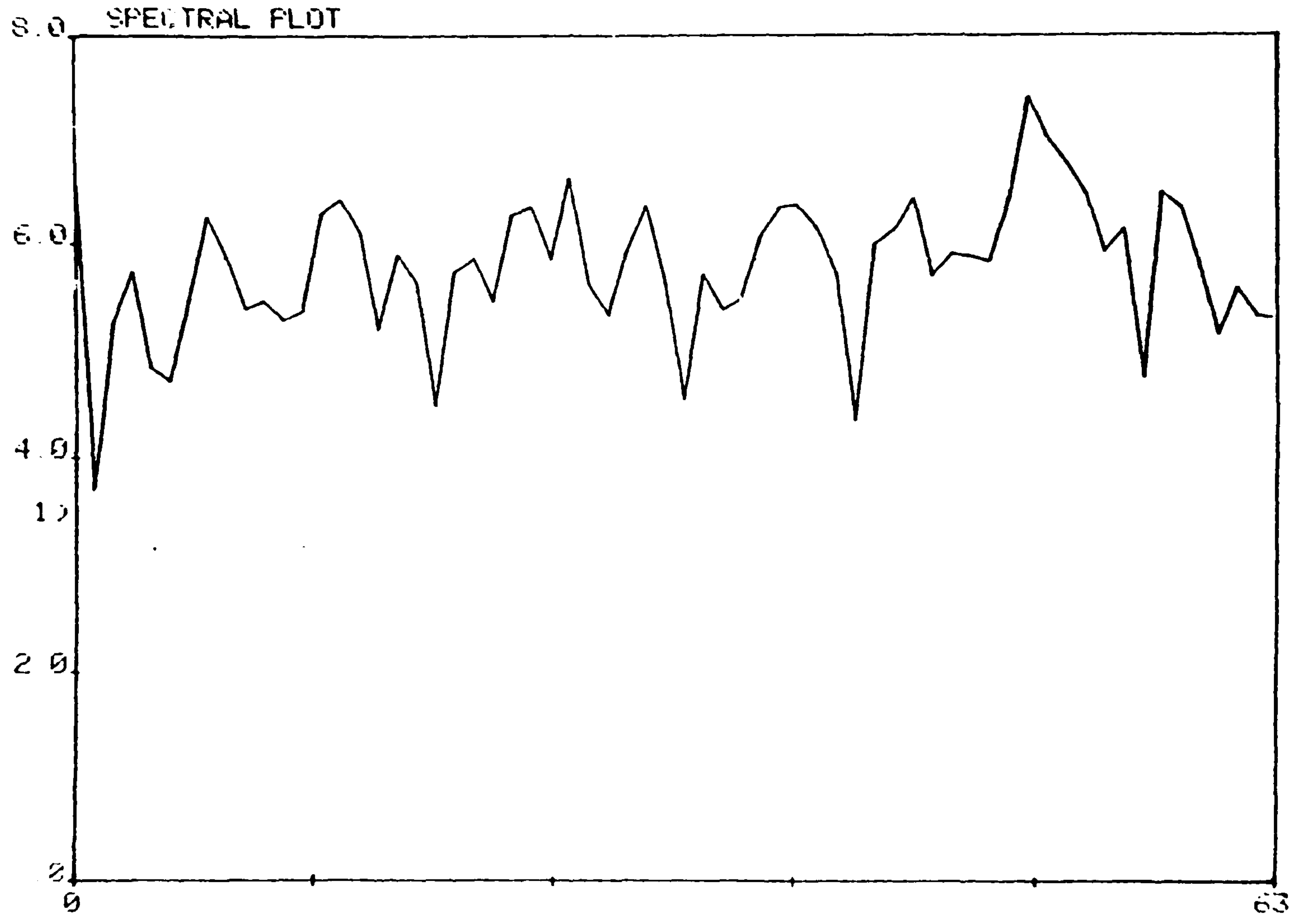


FIGURE 49. Spectral plot observation number 160, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

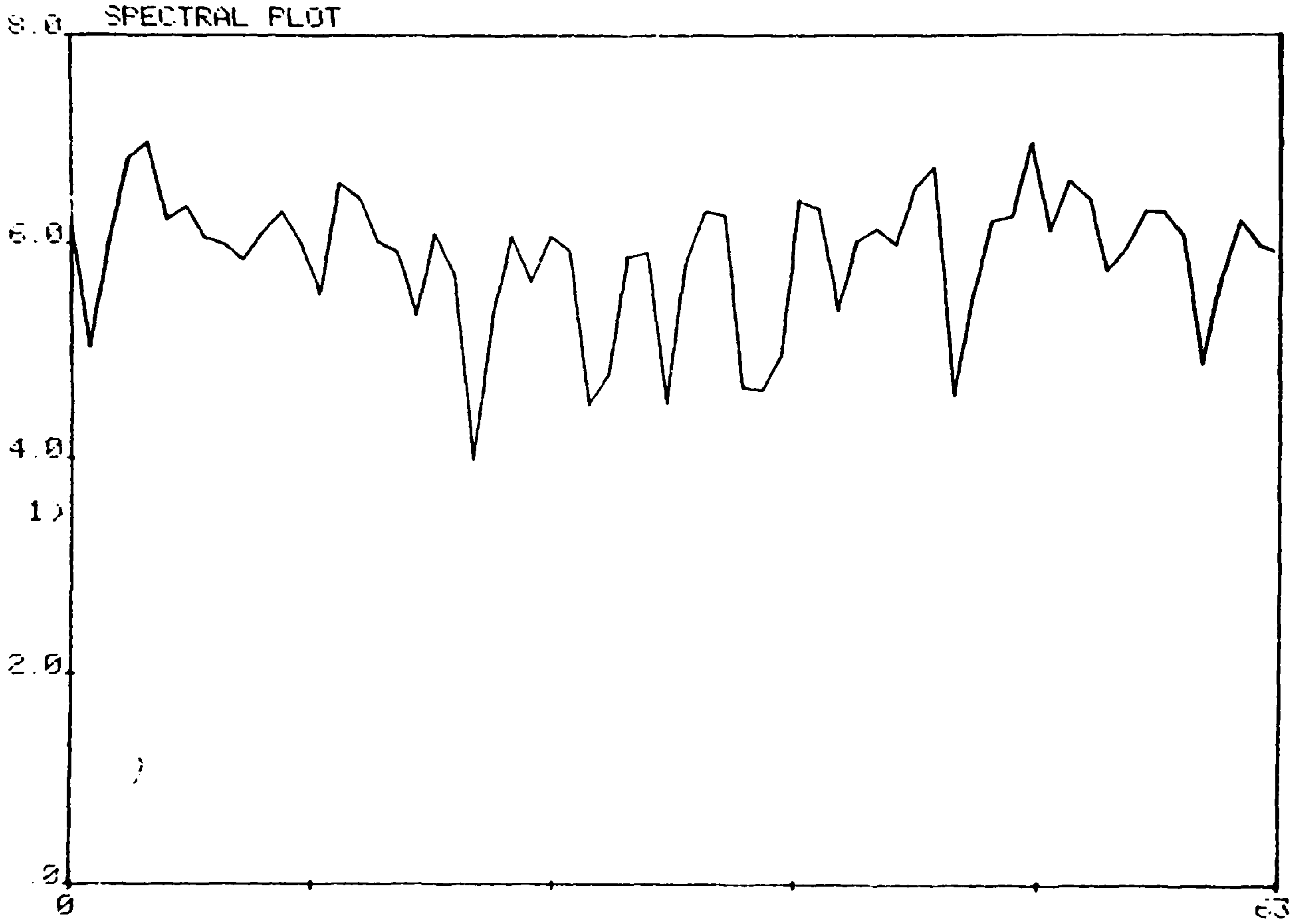


FIGURE 50. Spectral plot observation number 170, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

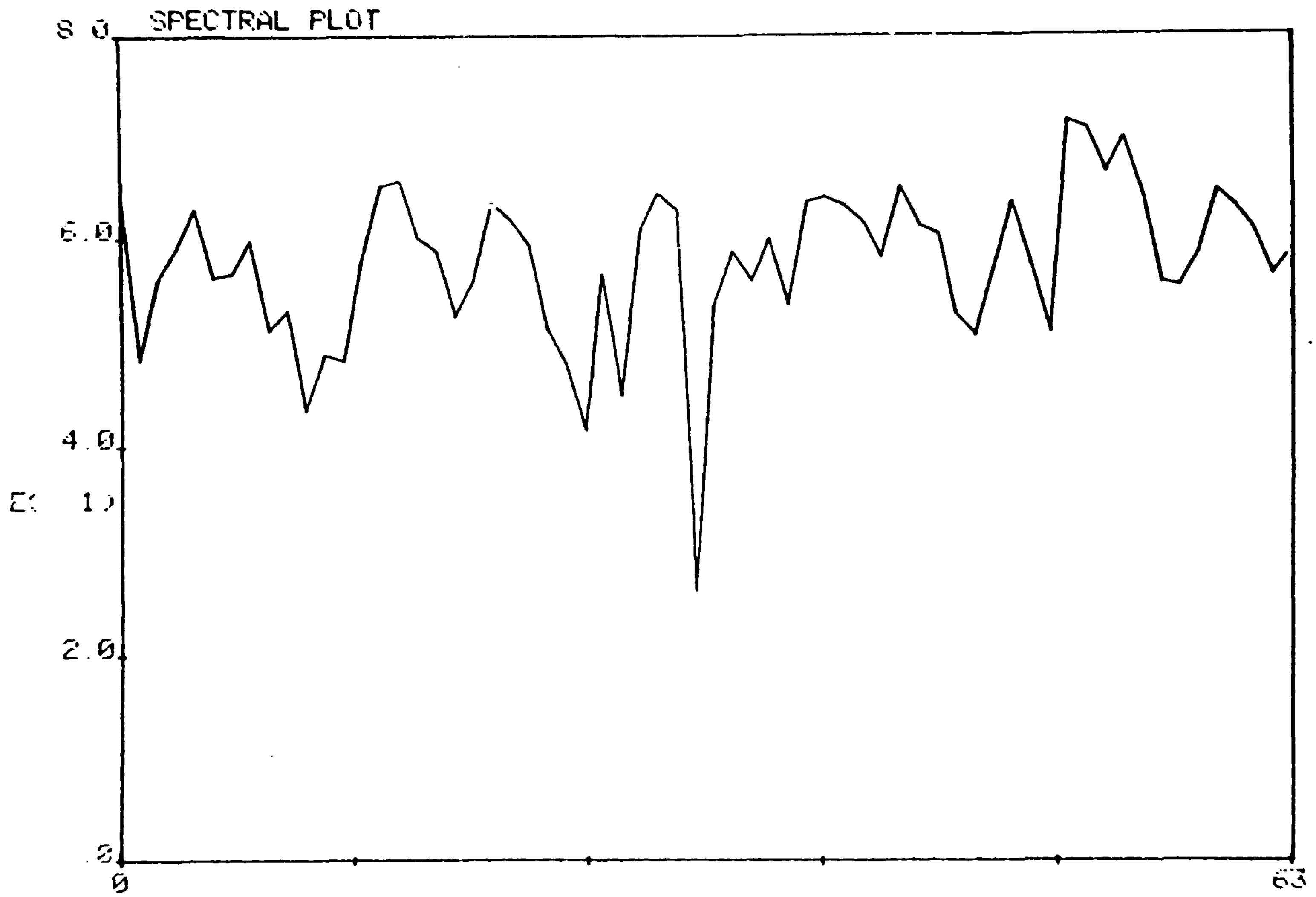


FIGURE 51. Spectral plot observation number 180, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

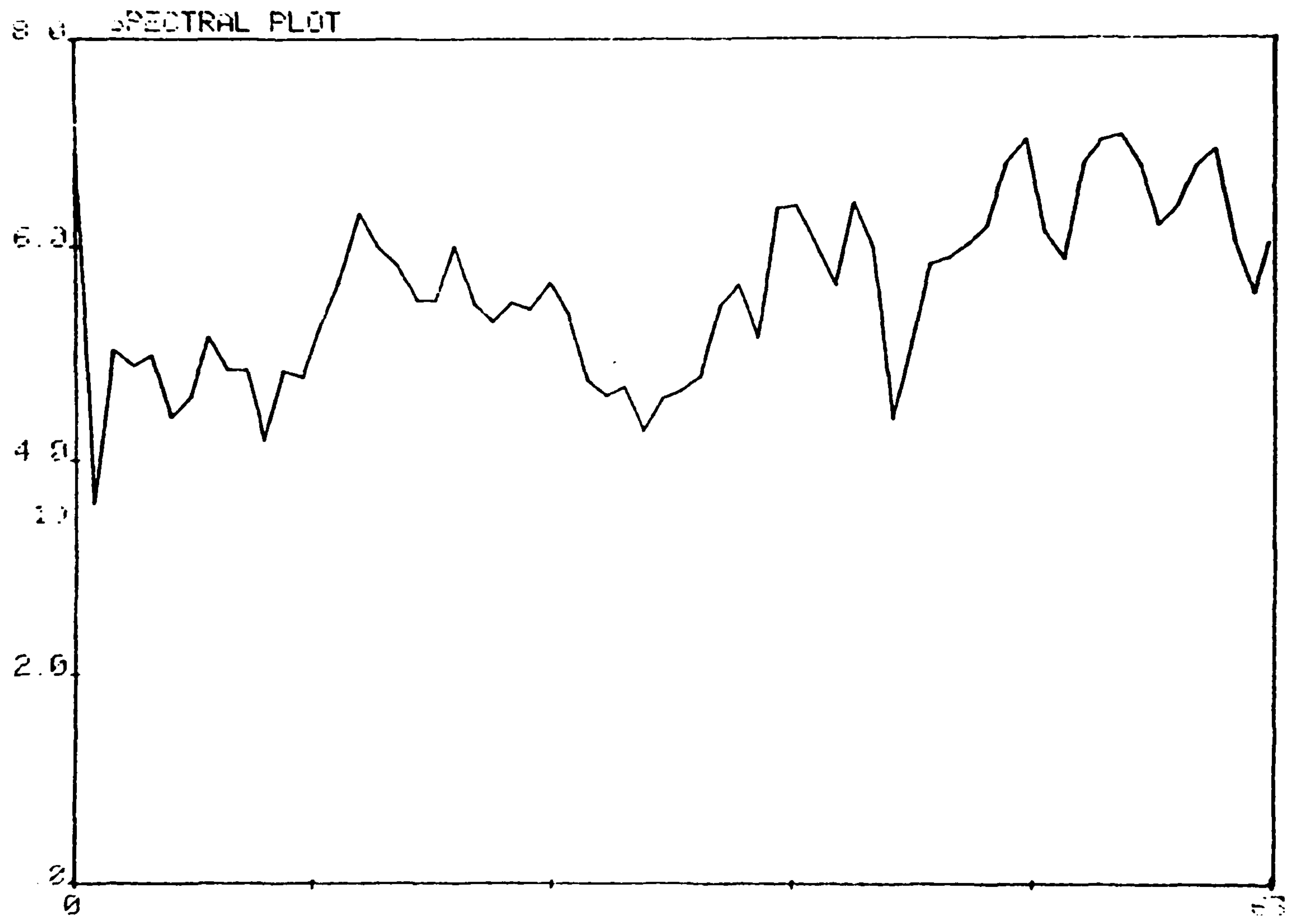


FIGURE 52. Spectral plot observation number 190, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

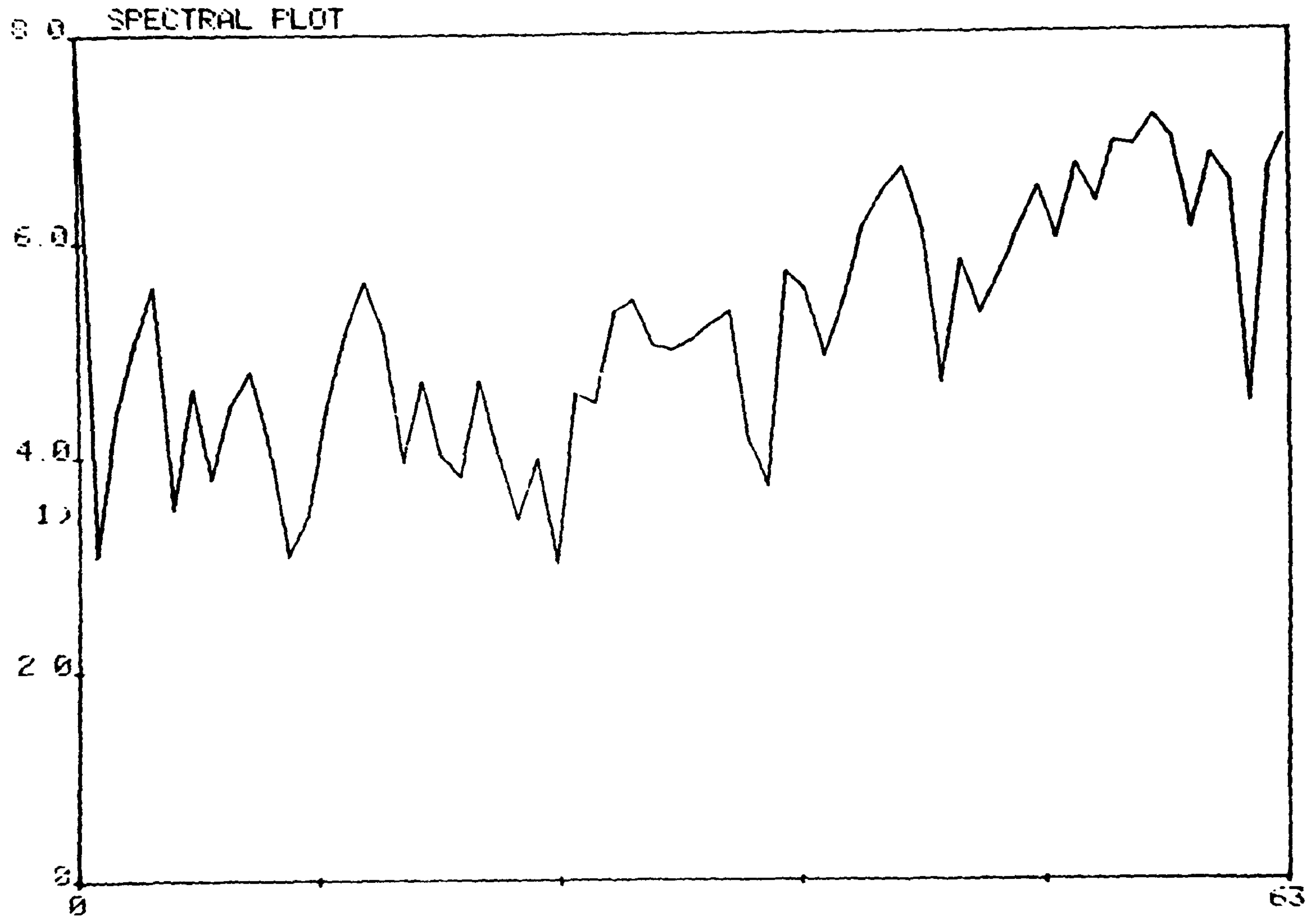


FIGURE 53. Spectral plot observation number 195, 128 point overlapping Hamming window, 10dB de-emphasis to 500HZ, 10dB pre-emphasis from 500HZ, energy normalized.

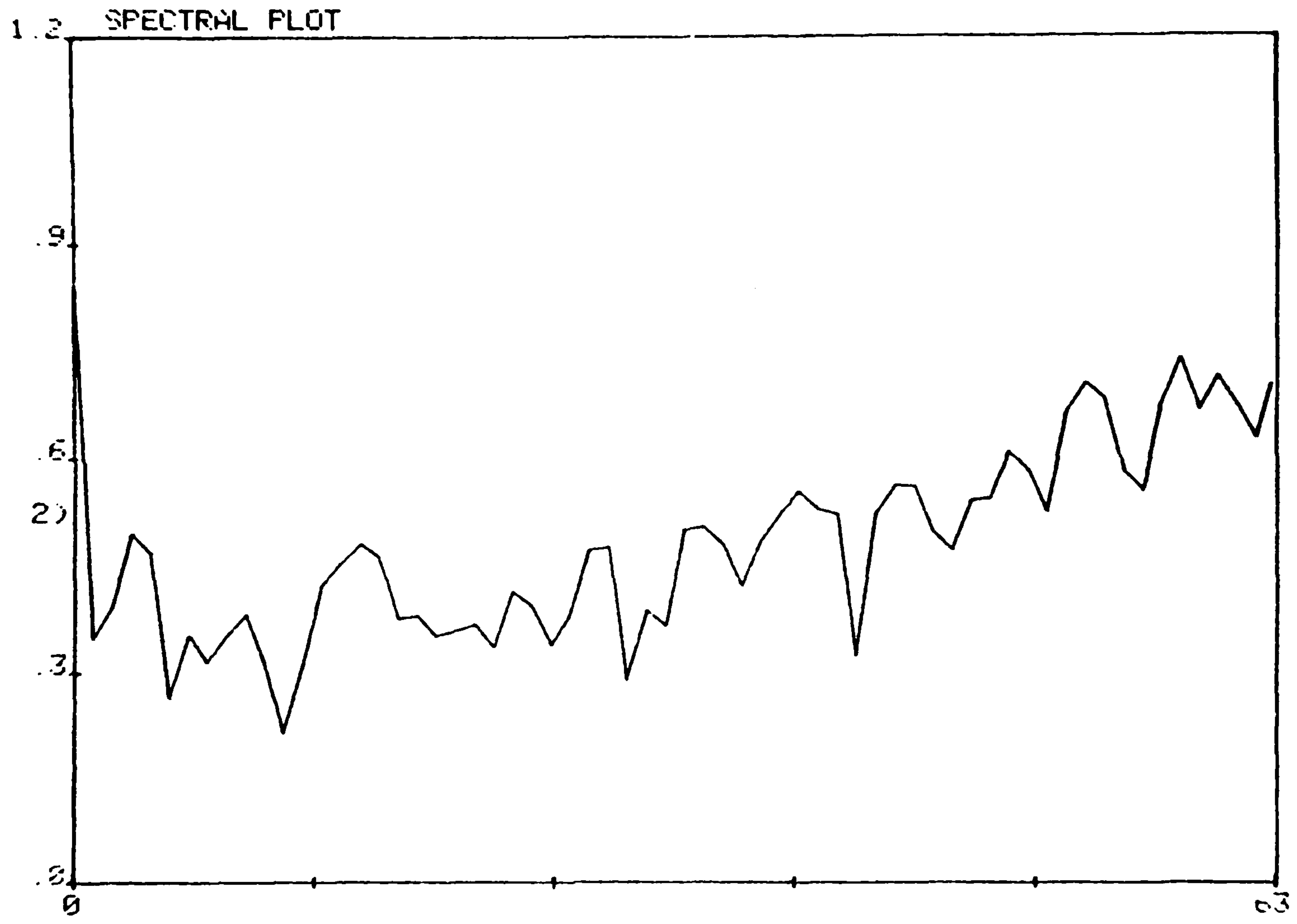


FIGURE 54. Spectral plot observation number 200, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

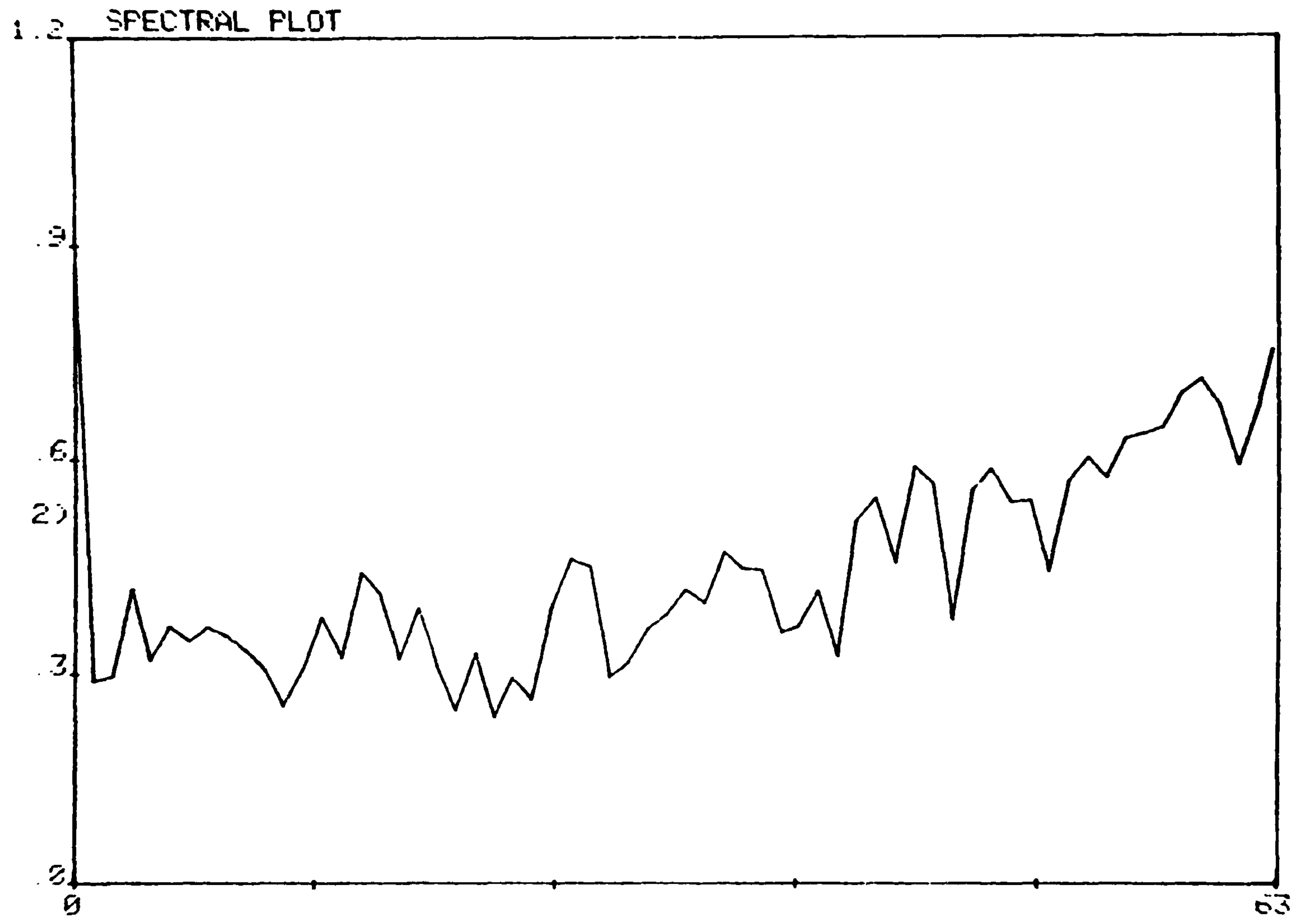


FIGURE 55. Spectral plot observation number 201, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

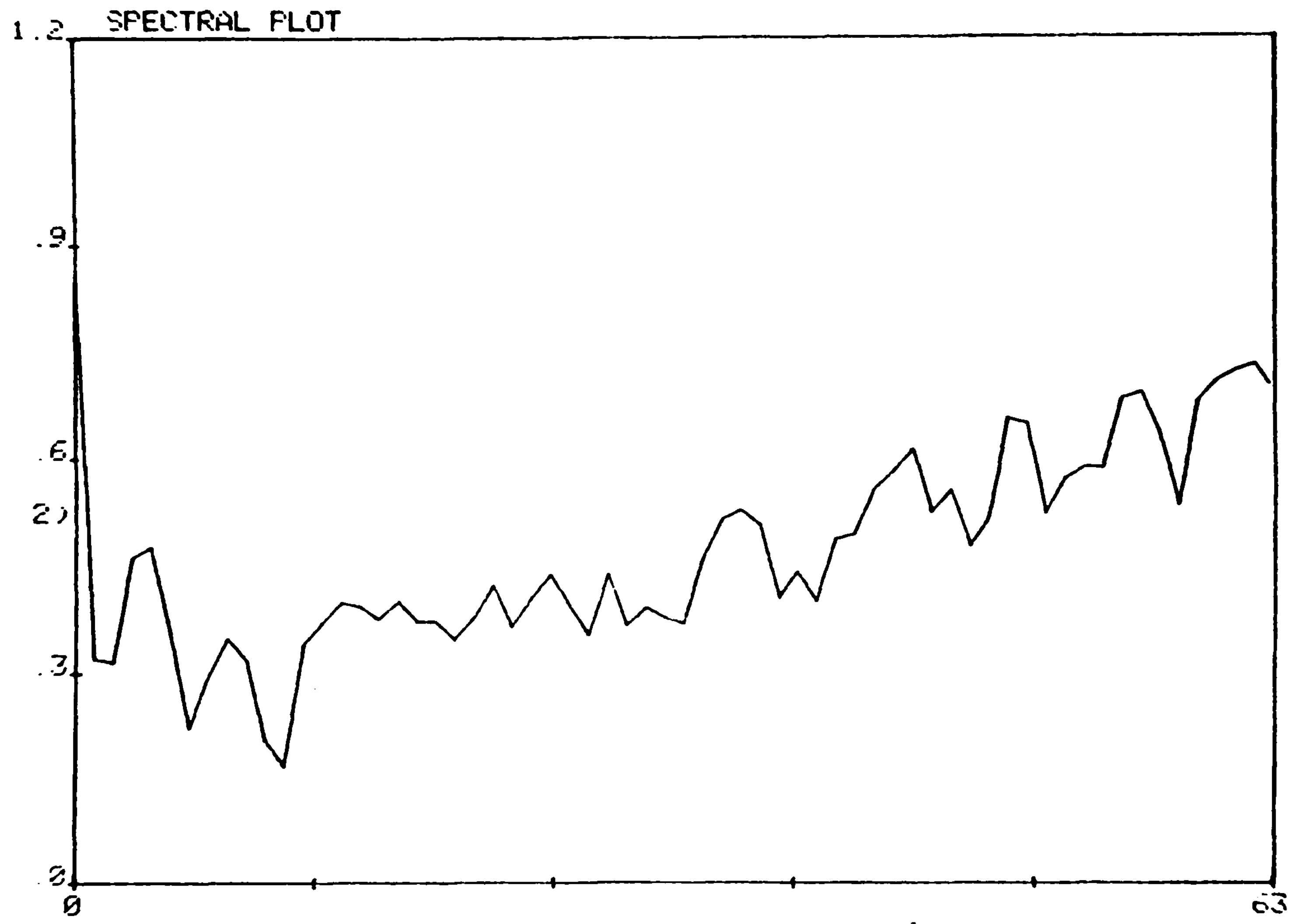


FIGURE 56. Spectral plot observation number 202, 128 point overlapping Hamming window, 10dB de-emphasis to 300HZ, 10dB pre-emphasis from 500HZ, energy normalized.

feature space will cluster for unvoiced sounds and noise sources. It can be argued, however, that success of a scheme for detecting phonetic units in noise will depend not so much on an energy threshold, but on how well phonets representing noise and speech separate, and on noise cancellation techniques employed. Several clustering algorithms (Ref 13) will need to be studied, and perhaps implemented, to quantify phonet detection in noise. A cluster algorithm is presented in the next section. But it seems too early, at this point, to settle on a threshold criterion such as Seelandt's.

A reliable discriminant between unvoiced speech and noise may be one based on clustering properties of observations about selected phonets. We would not expect distances from a white noise background model to cluster, but nonwhite noise may.

The distribution of similar observations in several speech segments is plotted in Figures 57-68. In each plot, M2 distances are plotted from a particular observation to each observation in the speech file. The smallest five and the largest distances are plotted along with the energy of the particular observation. The position along the horizontal axis at which a distance is plotted corresponds to the number of the observation to which that distance was computed. One of the minimum distances will be zero corresponding to a perfect match. The energy value is plotted in position zero.

Figures 57 and 58 plot the behavior of two observations from the nonspeaker region before the word "five" and illustrate the lack of clustering characteristics of low energy speakerless regions. Figures 63-65 plot distance choices from observations in the speakerless segment between the words "five" and "six" and again show the typical lack of clustering.

Figures 59 and 60 plot distance choices for observations in the "f" segment and in the "fi" segment, respectively. Observations from the "f" sound of this speaker did not cluster well; but, as the voiced "i" sound came in, distance choices clustered much tighter. Characteristic of the tight cluster in high energy vowel sounds is the plot in Figure 62. We were surprised to find that distance choices from observations in "s" segments clustered tightly. Figures 66, 67, and 68 plot choices from observations in the "s" sound in the word "six." From a spectrogram of the speech segment, we could see that these observations precede a strong "i" influence, and spectral plots from the region, Figures 54, 55, and 56, do not show the characteristic "i" structure. These are "s" sounds of moderate energy content which cluster.

Thus far in this chapter, we have presented the Acoustic Analyzer and illustrated observations and distances computed using values we have assigned to six parameters. We have discussed these parameters in both

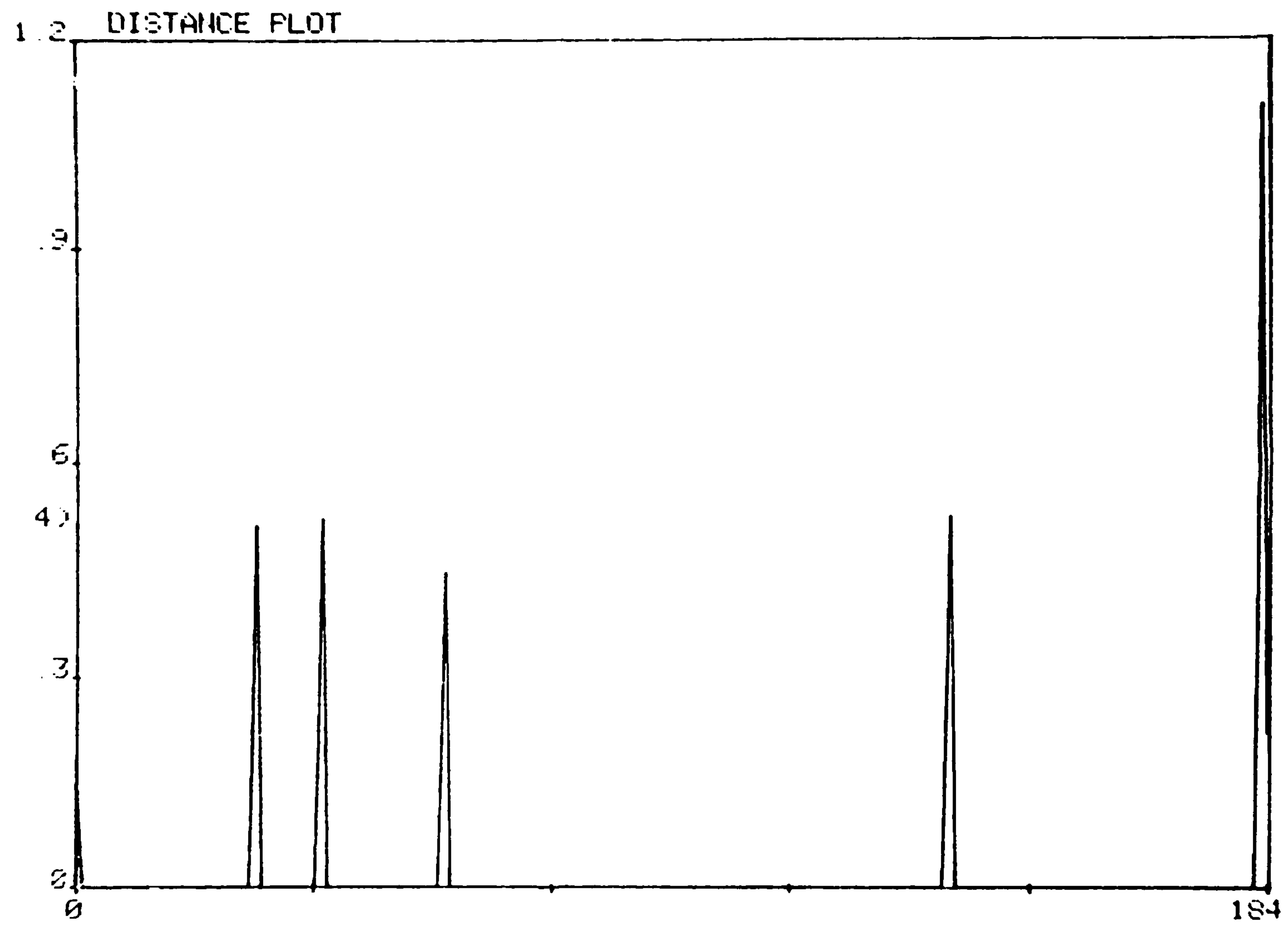


FIGURE 57. Five-Best Distances. Plot of observation energy in ordinate position zero, maximum and five-smallest distances from observation number 21, M2 rule.

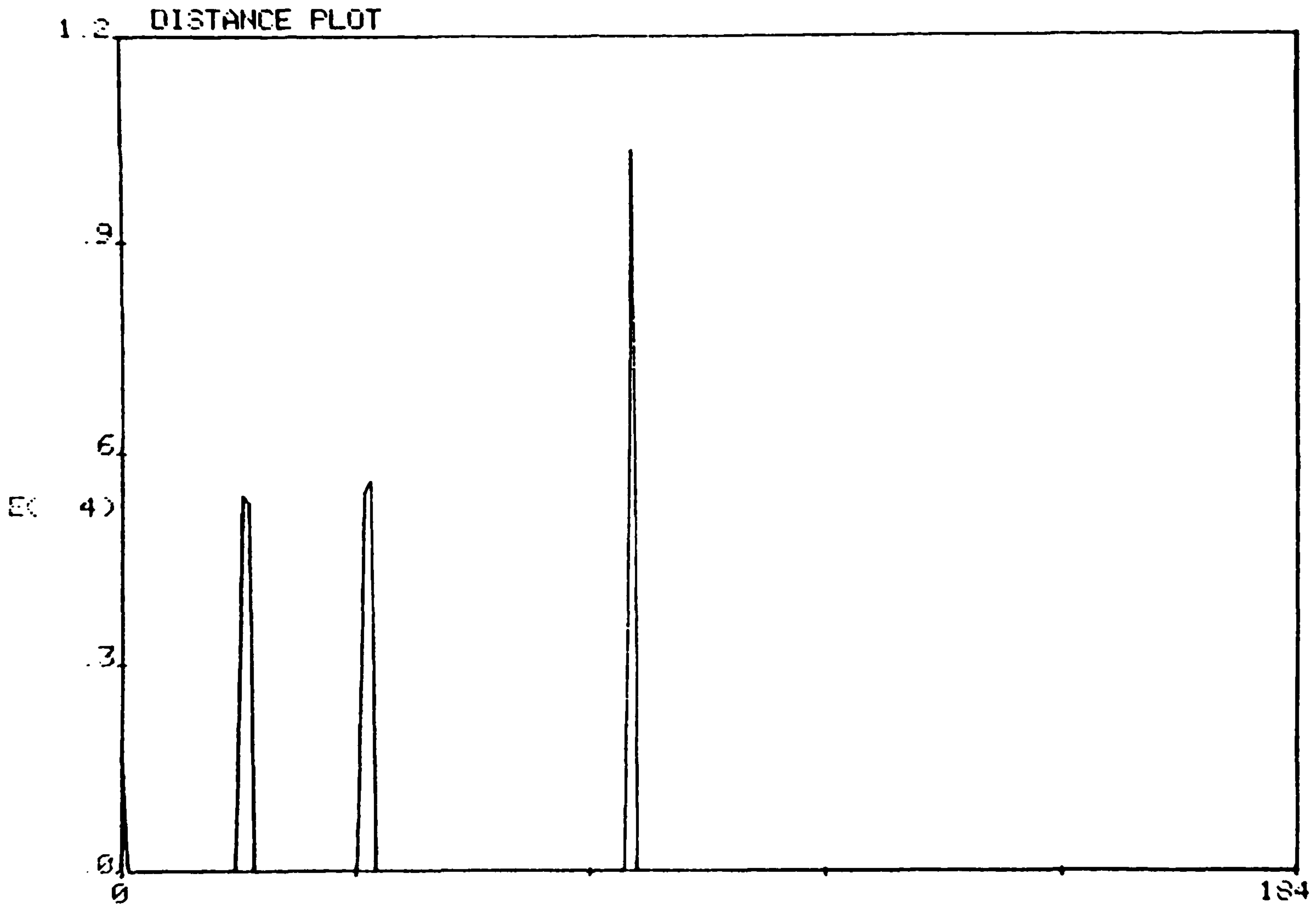


FIGURE 58. Five-Best Distances. Plot of observation energy in ordinate position zero, maximum and five-smallest distances from observation number 31, M2 rule.

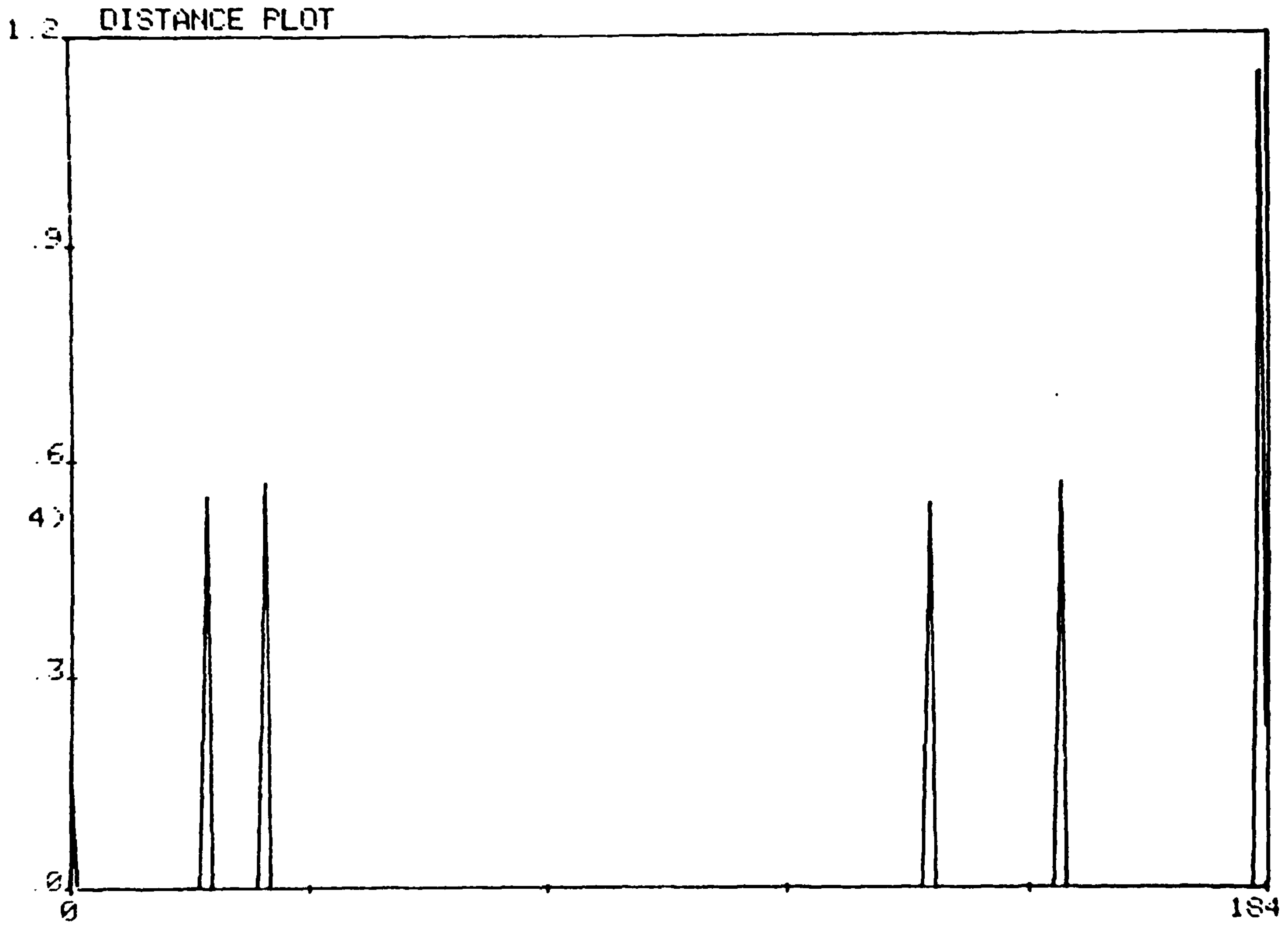


FIGURE 59. Five-Best Distances. Plot of observation energy in ordinate position zero, maximum and five-smallest distances from observation number 81 , M2 rule.

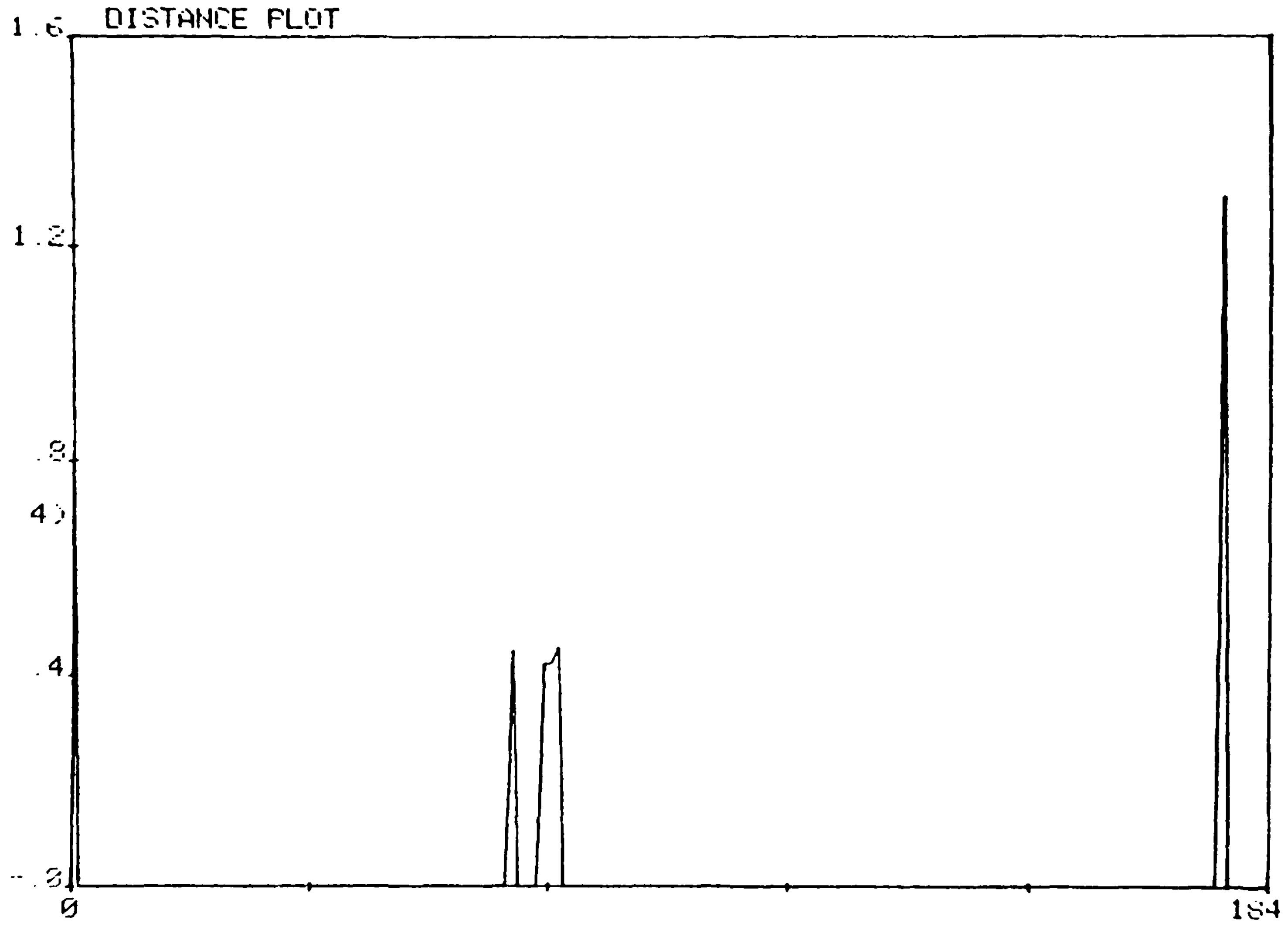


FIGURE 60. Five-Best Distances. Plot of observation energy in ordinate position zero, maximum and five-smallest distances from observation number 91, M2 rule.

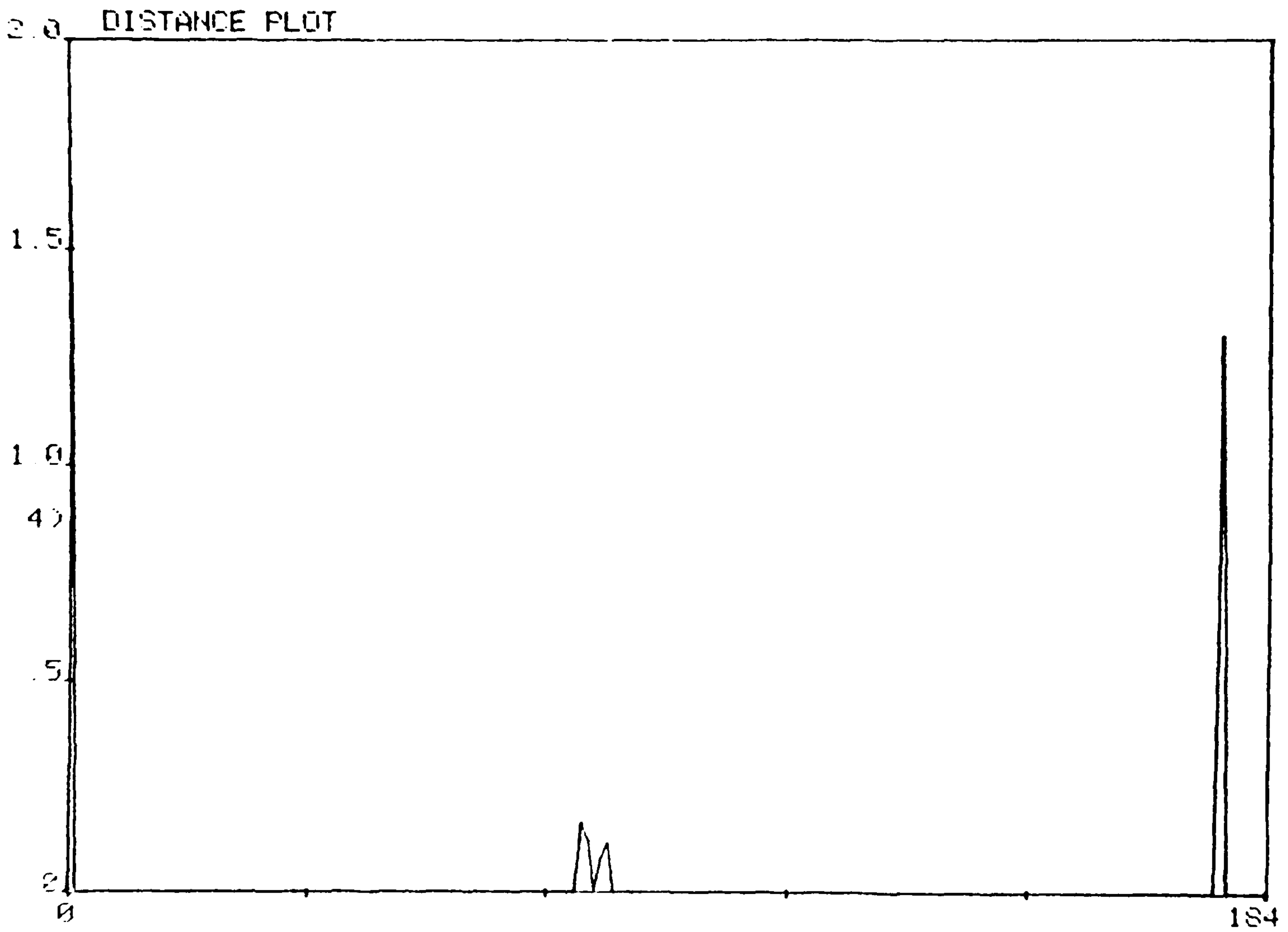
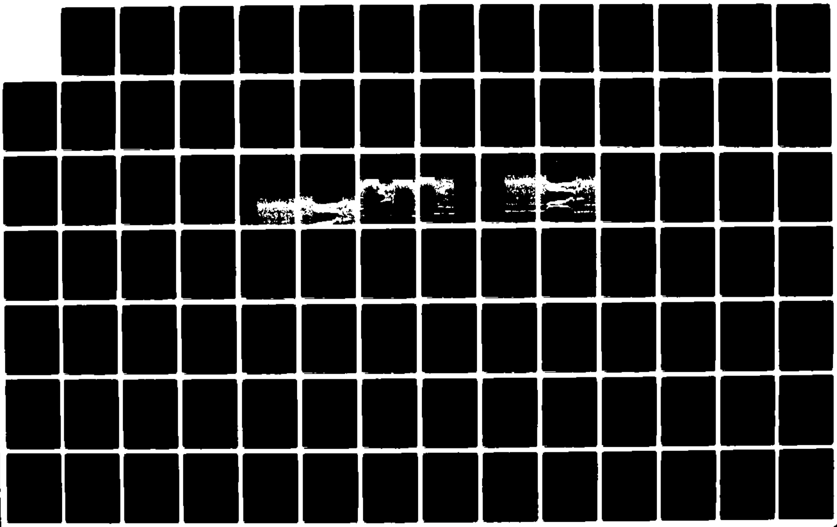


FIGURE 61. Five-Best Distances. Plot of observation energy in ordinate position zero, maximum and five-smallest distances from observation number 101, M2 rule.

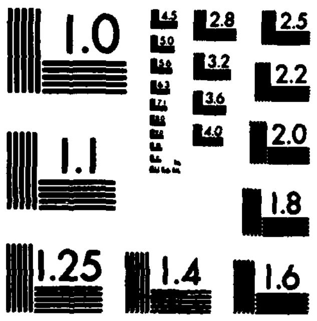
AD-A125 360 COMPUTER RECOGNITION OF PHONETS IN SPEECH(U) AIR FORCE INST OF TECH WRIGHT-PATTERSON AFB OH SCHOOL OF ENGINEERING D L MARTIN DEC 82 AFIT/GE/EE/82D-46

2/3

UNCLASSIFIED F/G 9/2 NL



M-2



MICROCOPY RESOLUTION TEST CHART
NATIONAL BUREAU OF STANDARDS-1963-A

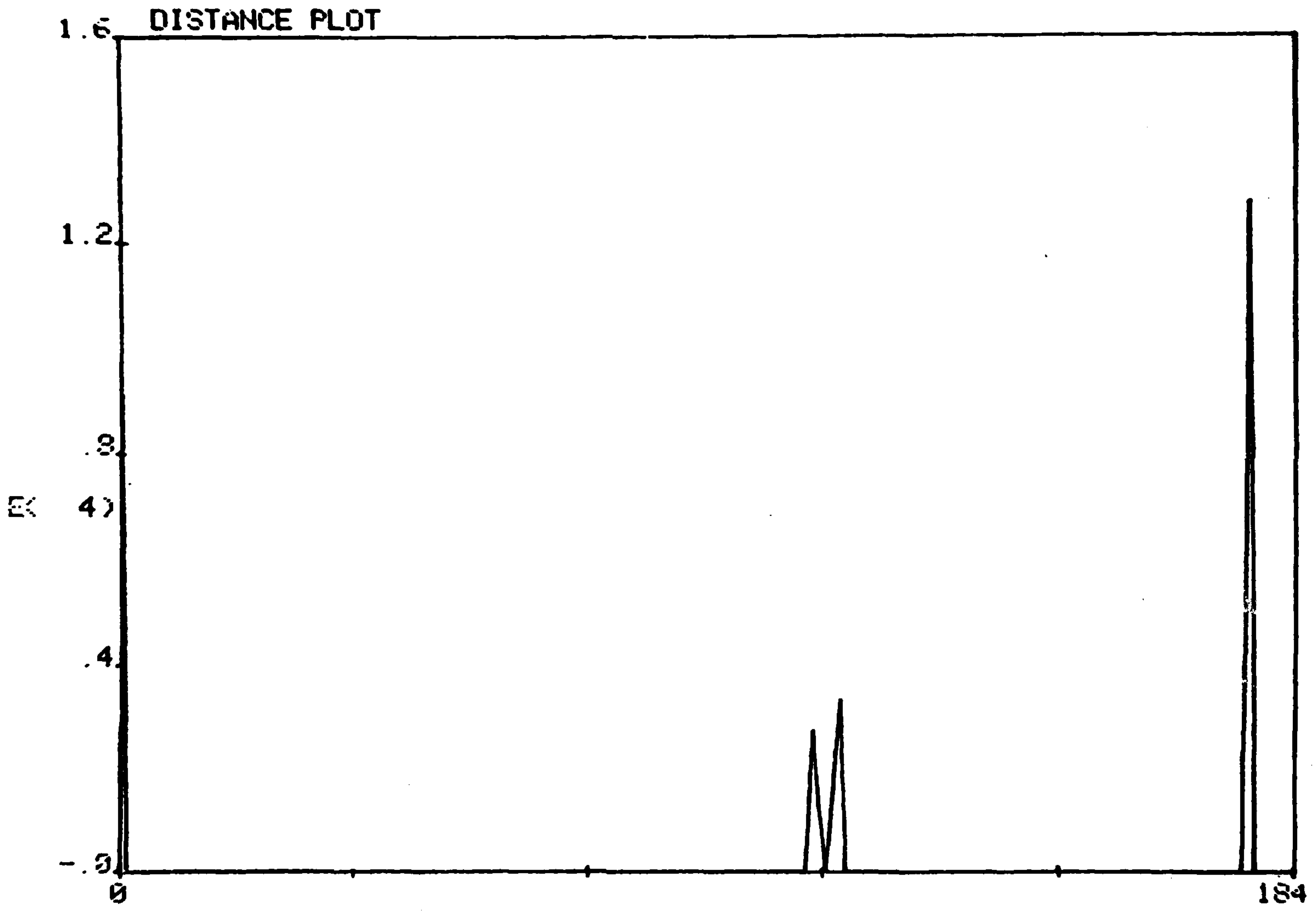


FIGURE 62. Five-Best Distances. Plot of observation energy in ordinate position zero, maximum and five-smallest distances from observation number 131, M2 rule.

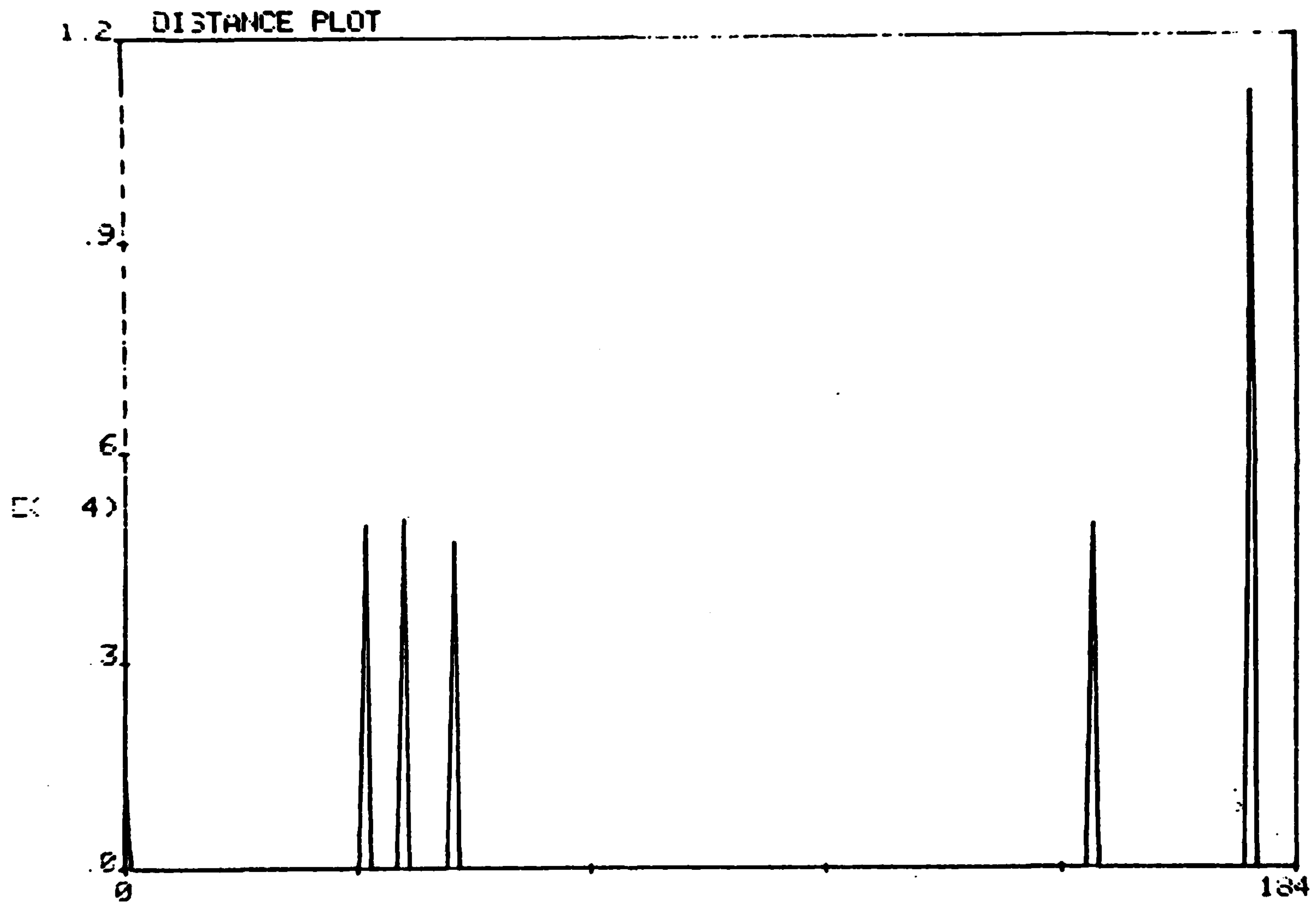


FIGURE 63. Five-Best Distances. Plot of observation energy in ordinate position zero, maximum and five-smallest distances from observation number 171 , M2 rule.

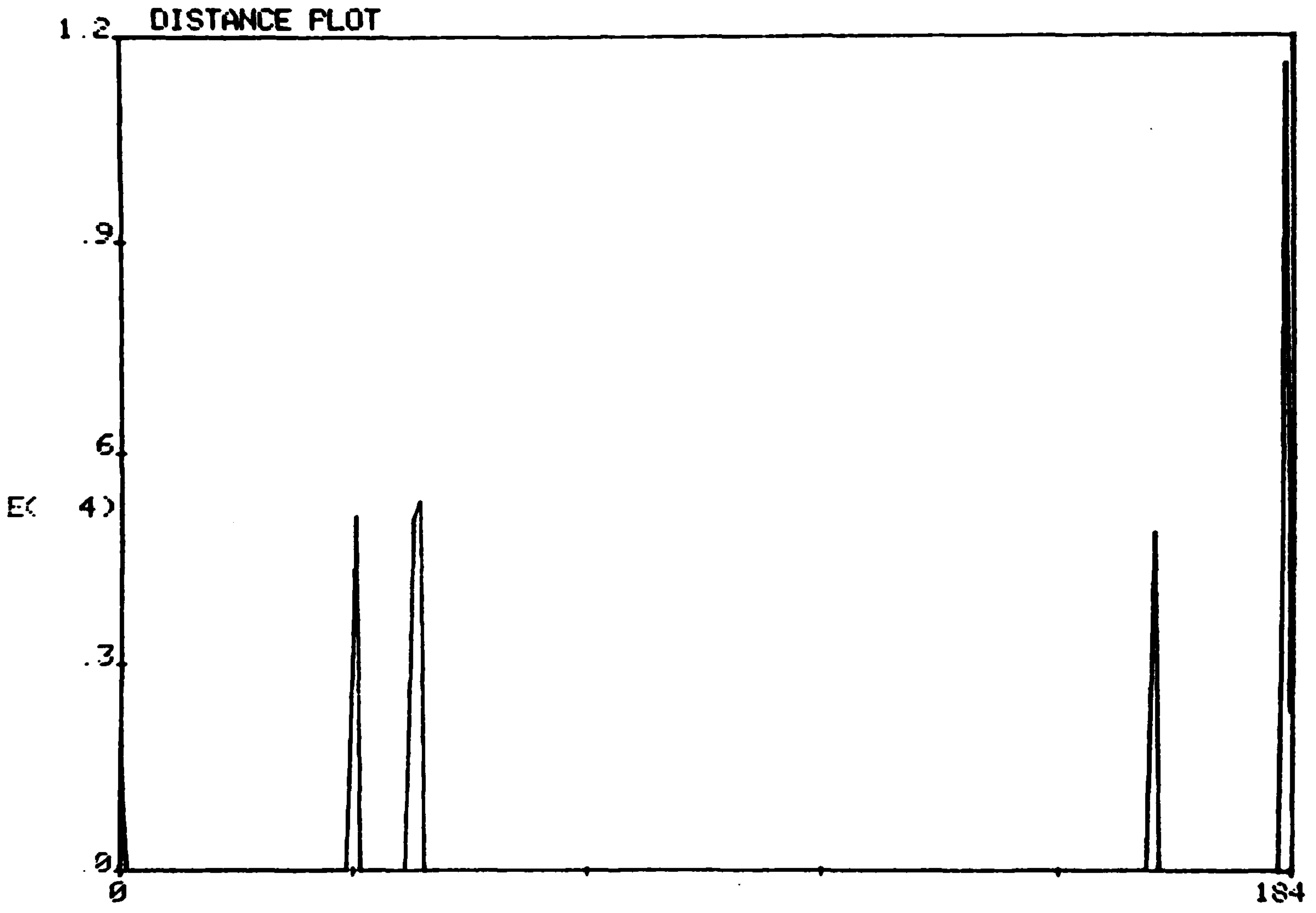


FIGURE 64. Five-Best Distances. Plot of observation energy in ordinate position zero, maximum and five-smallest distances from observation number 181, M2 rule.

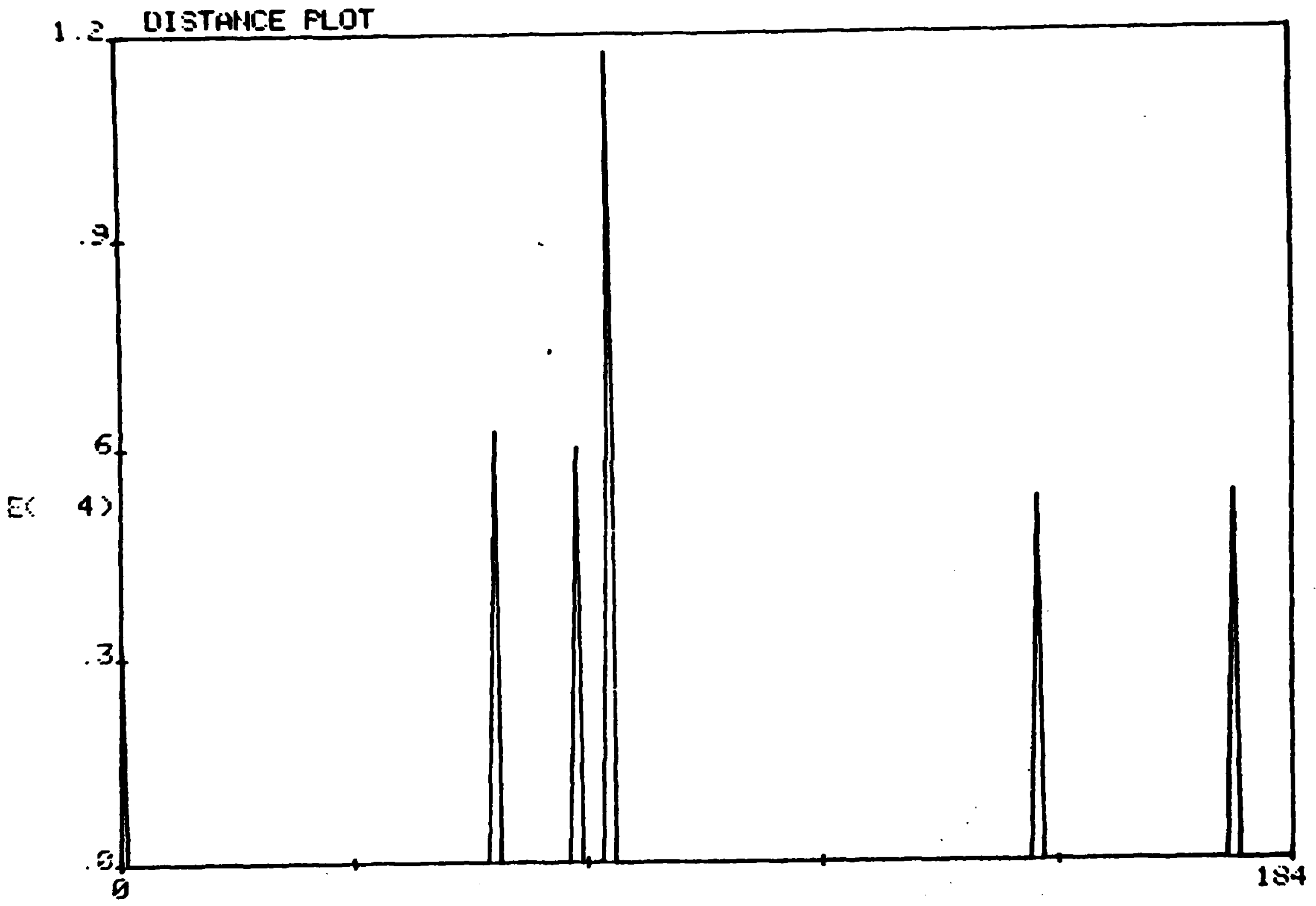


FIGURE 65. Five-Best Distances. Plot of observation energy in ordinate Position zero, maximum and five-smallest distances from observation number 191 , M2 rule.

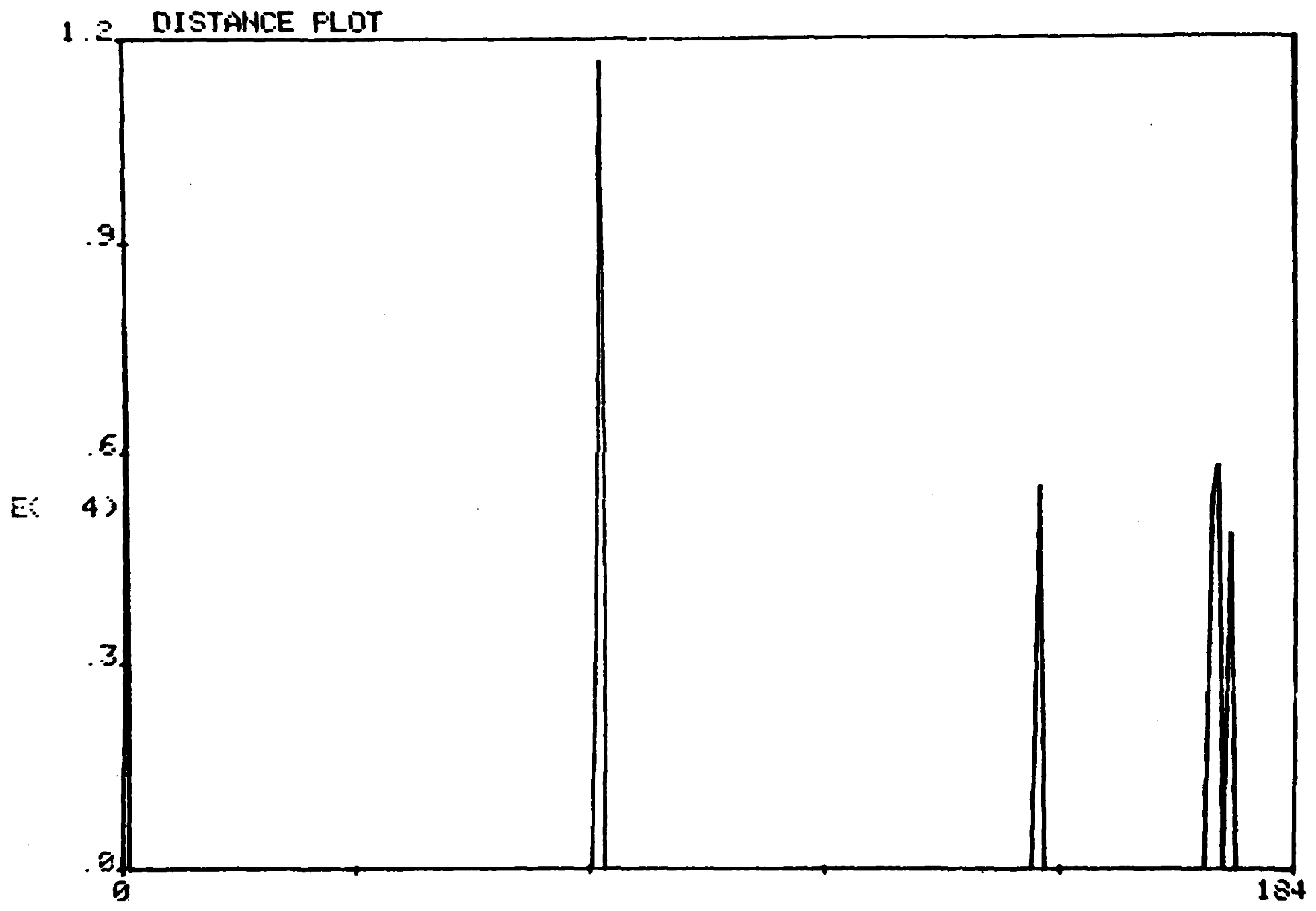


FIGURE 66. Five-Best Distances. Plot of observation energy in ordinate position zero, maximum and five-smallest distances from observation number 195, M2 rule.

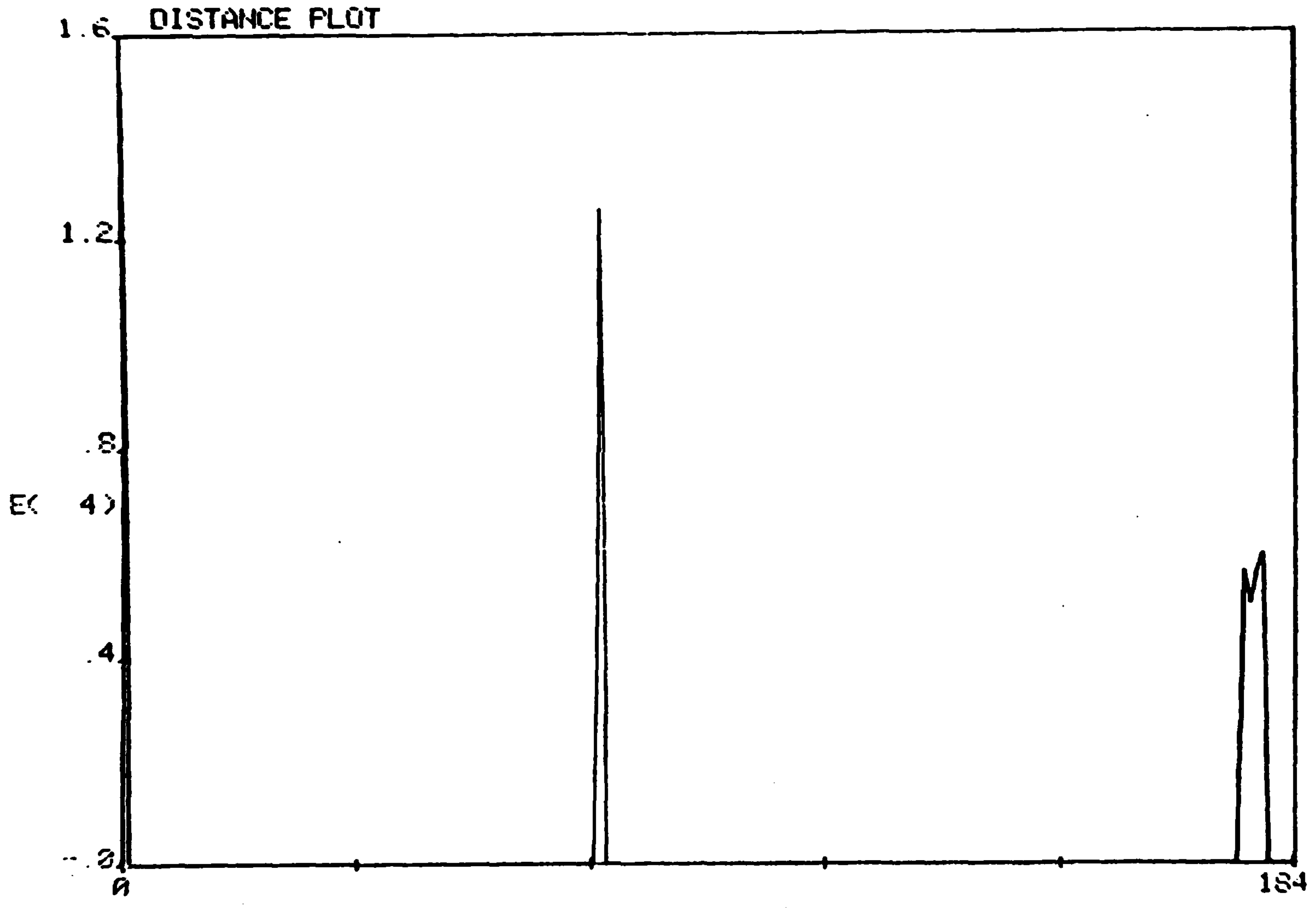


FIGURE 67. Five-Best Distances. Plot of observation energy in ordinate position zero, maximum and five-smallest distances from observation number 200 , M2 rule.

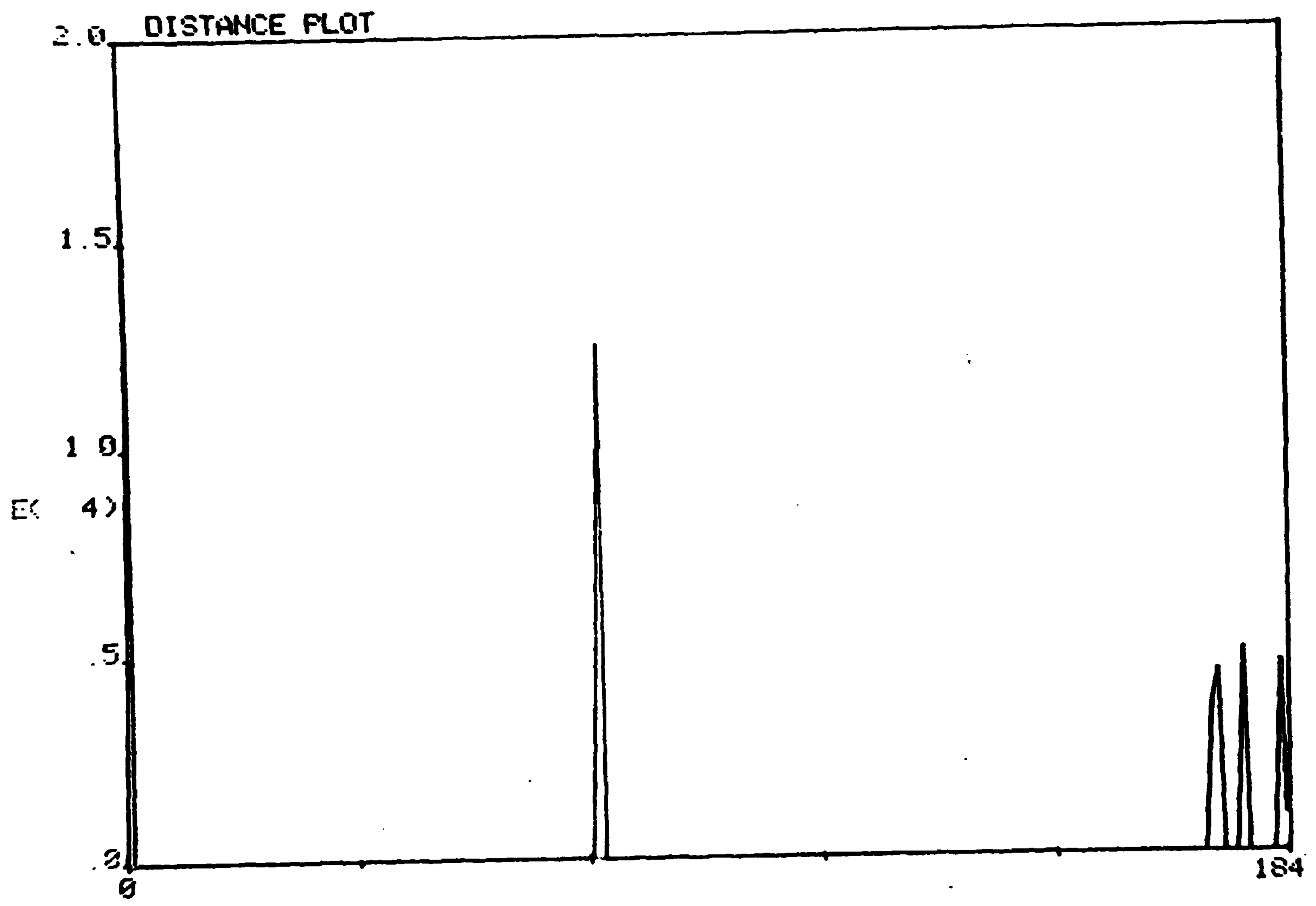


FIGURE 68. Five-Best Distances. Plot of observation energy in ordinate position zero, maximum and five-smallest distances from observation number 202 , M2 rule.

this and the previous chapter. We saw that adjacent observations may lie approximately the same distance from neighboring observations in segments of speech which contain high-to-moderate energy and pronounced formant structure. We saw this behavior for voiced vowels and for the "s" sound. We conclude from this observed behavior that observations generated from speech may cluster (Ref 13) in feature space. That is, it should be possible to classify observations without a priori knowledge based on a distance measure computed between them. In the next section, we describe an algorithm designed to do just that.

CLUSTER ALGORITHM:

In the previous section, we illustrated observations and their behavior under a distance measure. In this section, we describe an adaptive algorithm designed to cluster feature space into classes of observations. Classification is made on the basis of a decision rule without the benefit of a priori knowledge. The algorithm is adaptive in that a supervisor controls the modification of class descriptions in response to the classification results.

Pal, et. al., described an adaptive model (Ref 6) for computer recognition of vowel sounds using the first three formants as features. Their method uses a training procedure for self-supervised learning and a maximum value of fuzzy membership function as the basis of recognition. In

101

the remainder of this section, we will present an algorithm based on the above model, but modified to partition the space of observations in noise into clusters.

The model for our cluster algorithm is shown in Figure 69. It is redrawn from Reference 6. It uses a classifier which calculates a sufficient statistic between a set of phonets, $\underline{R}$, and an input observation, $\underline{X}$. The observation is assigned to the phonet class if the sufficient statistic is small enough; otherwise, the observation defines a new class.

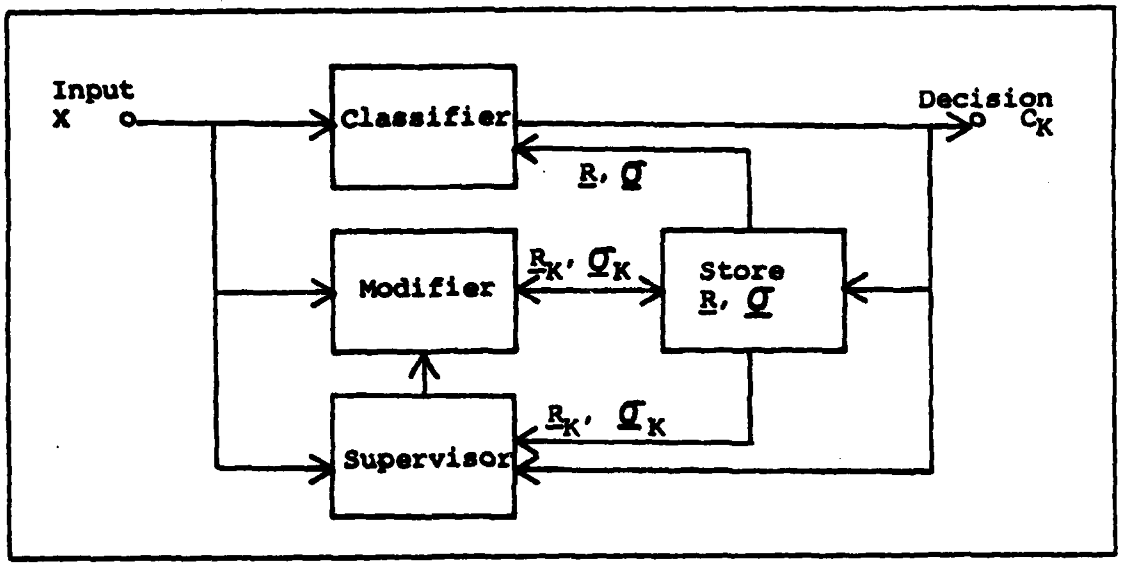


FIGURE 69. Model of an adaptive recognition scheme adapted from reference 6. The classifier assigns the input vector to the class to which it conforms most closely. The supervisor decides if the class descriptors should be modified.

102

Each sample, $\underline{X}_i$, in the sample space, Ω, is an observation computed by our Acoustic Analyzer. We assume that these samples have a central tendency about which they cluster. This central tendency is approximated by the ensemble average of all observations which cluster tightly about it.

The sample space is partitioned into a sequence of classes $\left\{A_i\right\}$. Each class is described by its central tendency, which we call a phonet; by its variance, which is the second central moment of all observations which cluster tightly about the central tendency; by an inner zone parameter which quantifies the adjective "tight" as applied to a cluster; and by an outer zone parameter which determines the outer boundary of the class. These class descriptors are calculated iteratively as the sample space is partitioned.

In this model, the supervisor assumes that observations have a central tendency about which they cluster. It also assumes that if an observation lies close enough to a central tendency, then the probability of misclassification is tolerable.

Accordingly, an annulus is constructed about each central tendency and each observation is compared to the boundaries of this annulus. If the observation is close enough to the central tendency to fall within the inner radius, then it is permitted to modify the description of that class. If it does not fall within the outer radius of any annulus, then a new class is created to which this observation is assigned.

In operation, a tentative classification of the i^{th} observation, $\underline{X}_i$, to the j^{th} class, A_j, is based on the minimum value of the sufficient statistic D(i, j); that is, tentatively assign $\underline{X}_i$ to class A_j if $D(i, j) = \underset{K}{\text{Min}}\ D(i, K)$.

The classification is made permanent on the basis of the two zone parameters, λ_1 and λ_2, according to the following rule:

(1) If $D(i, j) \leq \lambda_1$, then make the tentative classification permanent and modify the class descriptors.

(2) If $\lambda_1 < D(i, j) \leq \lambda_2$, then make the tentative classification permanent but do not modify the class descriptors.

(3) If $\lambda_2 < D(i, j)$, then delete the tentative classification, creat~ a new class, and assign $\underline{X}_i$ as the phonet descriptor of the class.

These zone parameters λ_1 and λ_2 are called the inner and outer zone parameters, respectively. They control the classification process and are related to detection performance.

The zone parameters can be set in a number of ways. They could be set arbitrarily and varied until optimum values are found experimentally. Or they could be related

ɜ probability of misclassification and noise energy
ɜ speech signal.

Che other two class descriptors, the phonet and
ıce vectors, can be calculated iteratively. Suppose
ıence of observations $\{\underline{X}_i\}$. Let $\underline{X}_i \cdot \underline{X}_j$ denote the
ɔlication of these two vectors component-by-component;
ıs, let:

$$\underline{X}_i \cdot \underline{X}_j = (x_{i1}\, x_{j1},\ x_{i2}\, x_{j2},\ \ldots,\ x_{iN}\, x_{jN}) \qquad (7)$$

ɜt $\underline{X}_i/\underline{X}_j$ denote the division of these two vectors
ıent-by-component. That is, let:

$$\underline{X}_i/\underline{X}_j = (x_{i1}/x_{j1},\ x_{i2}/x_{j2},\ \ldots,\ x_{iN}/x_{jN}) \qquad (8)$$

Given the first t observations of the sequence $\{\underline{X}_i\}$,
ırst noncentral moment is:

$$\underline{R}_t = \frac{1}{t}\sum_{i=1}^{t} \underline{X}_i, \qquad (9)$$

cond noncentral moment is:

$$\underline{C}_t = \sum_{i=1}^{t} \underline{X}_i \cdot \underline{X}_i, \qquad (10)$$

ıe second central moment is:

$$\underline{S}_t = \frac{1}{t}\,\underline{C}_t - (\underline{R}_t \cdot \underline{R}_t). \qquad (11)$$

ɜsume another observation is to alter these class
ıptors. Then:

$$\underline{R}_{t+1} = \frac{t}{t+1}\,\underline{R}_t + \frac{1}{t+1}\,\underline{X}_{t+1} \qquad (12)$$

$$\underline{C}_{t+1} = \underline{C}_t + (\underline{X}_{t+1} \cdot \underline{X}_{t+1}) \qquad (13)$$

$$\underline{S}_{t+1} = \frac{1}{t+1}\underline{C}_{t+1} - (\underline{R}_{t+1} \cdot \underline{R}_{t+1}). \qquad (14)$$

These equations provide the means for rapid, iterative calculation of the class descriptors using the Array Processor.

The algorithm given by Pal, et. al. (Ref 6), did not specify how to partition the feature space into classes. It assumed class partitions and their descriptors as initial conditions. That is, an initial set of class descriptors was resident in memory at the start of their algorithm. The algorithm given below in step-by-step format does not assume that the sample space is partitioned initially. Rather, it partitions feature space into classes whose boundaries are determined by noise in the system, the sufficient statistic, and class membership variability.

<u>STEP</u>	<u>ACTION</u>
0	Choose a variance vector, $\underline{V}_o$, and zone parameters λ_1 and λ_2. One way to do this is to let V_N be the noise variance and $\underline{V}_o = \lambda_1 \underline{V}_N$. Then set $\lambda_2 > \lambda_1$ by some multiple. Assume zero-mean noise.
1	Read the first observation, $\underline{X}_1$. Assign $\underline{X}_1$ to class A_1. Describe A_1 by: $N_1 = 1$, $\underline{R}_1 = \underline{X}_1$, $\underline{C}_1 = \underline{V}_o + (\underline{X}_1 \cdot \underline{X}_1)$, $\underline{V}_1 = \underline{V}_o$
2	Read the second observation, $\underline{X}_2$. Compute the supervisory parameter between $\underline{X}_2$ and $\underline{R}_1$, that is, compute D(2,1).

```
STEP    ACTION
        If $D(2,1) \le \lambda_1$
        Then:  assign $\underline{X}_2$ to $A_1$
               modify description of $A_1$
        else:  assign $\underline{X}_2$ to $A_2$
               describe $A_2$

3       Read the third observation, $\underline{X}_3$.
        Choose $\min_K D(3, K)$ for $K = 1, 2$

        Say the minimum is at $K = 1$.
        If $D(3, 1) \le \lambda_1$
          Then:  assign $\underline{X}_3$ to class $A_1$
                 modify description of $A_1$
          else:  assign $\underline{X}_3$ to class $A_3$
                 describe $A_3$

•

•

•

M       Read m^th observation $\underline{X}_m$.
        Choose $\min_K D(M, K)$ for $K = 1, 2, \ldots, n$

        Say the minimum is at $K = 1$.
        If $D(M, 1) \le \lambda_1$
          Then:  assign $\underline{X}_m$ to class $A_1$
                 modify description of $A_1$
          else:  assign $\underline{X}_m$ to class $A_{n+1}$
                 describe class $A_{n+1}$

•

•

•
```

101

Continue in this way until M_u classes have been described. M_u is the largest number of classes tolerable and M_l is the smallest number of classes tolerable.

At this point, M_u classes have been described. Each class was created because the observation currently under consideration was not described adequately by any of the classes then in existence. Adequacy was defined in terms of class variance. Now, at most, M_u classes are to be permitted and one cannot be sure at this point that future observations will be described adequately by the current set of M_u classes. If a future observation is not described adequately, a new class will need to be created. Since processing limitations will intrinsically set some upper limit on M_u, the number of classes must be reduced to M_l, some number less than M_u. This reduction is to be accomplished in such a way as to increase class variance for some of the classes retained so that each class describes adequately more of the feature space. This is done by combining the most similar classes in pairs; that is, by treating the phonets as observations and computing the distance measure on the phonet set.

<u>STEP</u>	<u>ACTION</u>
.	
.	
.	
M+1	Read the first phonet $\underline{R}_1$. Compute the distance between $\underline{R}_1$ and every

STEP	ACTION
	other phonet, that is, D (1, K) for K = 2, ..., M_u. Choose $\underset{K\neq 1}{\text{Min}}$ D(1, K).
	Say the minimum is at K = 1.
	If D(1, 1) $\leq \lambda_1$
	Then: modify description of class A_1 using $\underline{R}_1$ as an observation
	else: do not modify A_1 descriptors
	Delete class A_1.
M+2	Read next phonet, $\underline{R}_2$.
	Choose $\underset{K\neq 2}{\text{Min}}$ D(2, K) for K = 3, ..., M_u
	Say minimum at K = 1.
	If D(2, 1) $\leq \lambda_1$
	Then: modify A_1 descriptors using $\underline{R}_2$
	else: do not modify A_1.
	Delete class A_2.
•	
•	
•	
M+r	Read phonet $\underline{R}_r$
	Choose $\underset{K\neq r}{\text{Min}}$ D(r, K) for K = r+1, ..., M_u
	Say miminum at K = 1
	If D(r, 1) $\leq \lambda_1$
	Then: modify description of A_1 using $\underline{R}_r$
	else: do not modify A_1
	Delete A_r.
•	
•	
•	

Continue on in this way until the number of classes is reduced to M_1. Renumber the classes A_1 through A_{M1}.

Once the phonet space is reduced to M_1 classes, then classification proceeds using the rule D(i, j) as was done above when the phonet vectors were classified. Now, however, the outer zone parameter is applied to the detection problem to determine if a new class should be added to the set $\{A_i\}$.

```
STEP      ACTION

  .
  .
  .
  t       Read observation $\underline{X}_n$.
          Choose $\min_K D(n, K)$ K = 1, ..., $M < M_u$
          Say minimum at K = 1.
          If $D(n, 1) < \lambda_1$
            Then:  assign $\underline{X}_n$ to class $A_1$
                   modify class descriptors for $A_1$
            else:  If $D(n, 1) \leq \lambda_2$
                      then:  assign $\underline{X}_n$ to class $A_1$
                             do not modify descriptors
                      else:  assign $\underline{X}_n$ to $A_{M+1}$
                             describe class $A_{M+1}$
  .
  .
  .
```

Continue in this way until $M+1 = M_u$. When $M+1 = M_u$, then reduce the phonet space until there are M_1 classes. Then classify more observations...

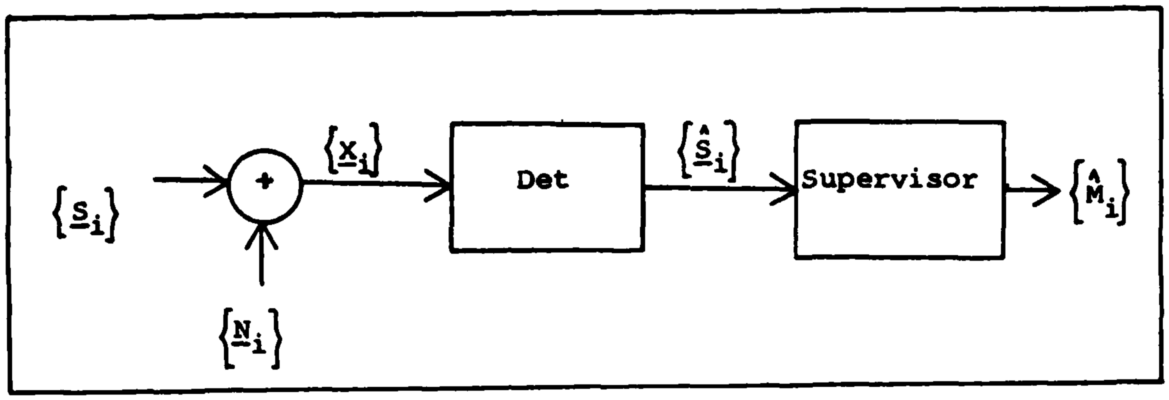


FIGURE 70. Model of the Supervisory Detection Process.

A model of the supervisory detection process is shown in Figure 70. The additive noise vector is assumed to have uncorrelated components, each distributed with zero-mean and variance V_i. Thus $\underline{X}_i = \underline{S}_i + \underline{N}_i$, its mean is $E[\underline{X}_i] = \overline{\underline{X}}_i = \underline{S}_i$, and its variance is:

$$V = \begin{pmatrix} V_1 & & & & 0 \\ & V_2 & & & \\ & & \cdot & & \\ 0 & & & \cdot & \\ & & & & \cdot \\ & & & & V_m \end{pmatrix} \qquad (15)$$

Wozencraft and Jacobs have shown that the optimum decision rule is to choose the class, A_i, to which the observation, $\underline{X}_i$, most probably belongs (Ref 14:212-214). That is, choose

$$A_j: \quad \underset{i}{\text{Max}} \; P\left\{S_i\right\} \; P_N \; (\underline{X} \,|\, \underline{S}_i) \tag{16}$$

where the set of a priori probabilities are $\left\{P\left\{\underline{S}_i\right\}\right\}$ and $P_N\ (\underline{X}\,|\,\underline{S}_i)$ is assumed to be the joint - M guassian density of $\underline{X}$ conditioned on $\underline{S}_i$. Melsa and Cohn give the density with mean $\underline{S}_i$ and variance V (Ref 15:69), and develop the multiple decision problem (Ref 15:113-116). We parallel their development.

The joint - M guassian density is:

$$P_N\ (\underline{X}\,|\,\underline{S}_i) = \frac{\exp\left\{-\frac{1}{2}\ (\underline{X} - \underline{S}_i)^T\ V^{-1}\ (\underline{X} - \underline{S}_i)\right\}}{\sqrt{\prod_{j=1}^{M} 2\pi v_j}} \tag{17}$$

The optimum decision rule, equation 16, is met when the logarithm of its terms is maximum, so we express our decision rule as:

$$A_j: \quad \underset{i}{\text{Max}} \; U_i - \frac{1}{2}\ (\underline{X} - \underline{S}_i)^T\ V^{-1}\ (\underline{X} - \underline{S}_j) \tag{18}$$

where:

$$U_i = \ln P\left\{\underline{S}_j\right\} - \frac{1}{2} \sum_{i=1}^{M} \ln\ (2\pi v_i) \tag{19}$$

Assuming equally likely a priori probabilities, then $U_i = U_j \;\forall\; i, j$ and our decision rule reduces to:

$$A_j: \quad \underset{i}{\text{Min}} \; (\underline{X}_i - \underline{S}_i)^T\ V^{-1}\ (\underline{X} - \underline{S}_i) \tag{20}$$

We define the sufficient statistic:

$$D(i, j) = (\underline{X}_i - \underline{S}_j)^T V^{-1} (\underline{X}_i - \underline{S}_j) \quad (21)$$

and write the decision rule as:

$$A_j: \ \underset{K}{\text{Min}}\ D(i, K). \quad (22)$$

Our algorithm approximates $\underline{S}_i$ as the ensemble average of all observations which it is sufficiently certain belongs to the class A_i, and the variance V_i as the ensemble variance of those observations.

The probability of error is the probability that there exists at least one class, A_K, given $\underline{S}_j$, for which $D(i, K) < D(i, j)$. That is, we let E_K denote the event:

$$E_K = \left\{W: \ D(i, K) < D(i, j) \middle| \underline{S}_j\right\} \quad (23)$$

where the statistics are, themselves, random variables and their dependence on the sample point, W, is suppressed for convenience.

We apply the union bound as given by Melsa and Cohn (Ref 15:115) to upper bound the probability of error. Say there are L classes. Then we say:

$$P\left\{E \middle| \underline{S}_j\right\} < \sum_{\substack{K=1 \\ K \neq j}}^{L} P\left\{E_K \middle| \underline{S}_j\right\}. \quad (24)$$

The task is to find $P\left\{E_K \middle| \underline{S}_j\right\} \ \forall\ K \neq j,\ K \leq L$.

The event E_K occurs whenever, given $\underline{S}_j$,

$$D(i, K) < D(i, j) \quad (25)$$

$$(\underline{X}_i - \underline{S}_K)^T V^{-1} (\underline{X}_i - \underline{S}_K) < (\underline{X}_i - \underline{S}_j)^T V^{-1} (\underline{X}_i - \underline{S}_j) \quad (26)$$

$$\underline{X}_i^T V^{-1} X_i - \underline{S}_K^T V^{-1} \underline{X}_i - \underline{X}_i^T V^{-1} \underline{S}_K + \underline{S}_K^T V^{-1} S_K$$
$$< \underline{X}_i^T V^{-1} \underline{X}_i - \underline{S}_j^T V^{-1} \underline{X}_i - \underline{X}_i^T V^{-1} \underline{S}_j + \underline{S}_j^T V^{-1} \underline{S}_j$$
$$\underline{S}_K^T V^{-1} \underline{S}_K - \underline{S}_j^T V^{-1} \underline{S}_j < 2\underline{X}_i^T V^{-1} (\underline{S}_K - \underline{S}_j) \tag{27}$$
$$2\underline{X}_i^T V^{-1} (\underline{S}_K - \underline{S}_j) > \underline{S}_K^T V^{-1} \underline{S}_K^T - \underline{S}_j^T V^{-1} \underline{S}_j.$$

Using $\underline{X}_j = \underline{S}_j + \underline{N}_j$, we have:

$$2(\underline{S}_j + \underline{N}_j)^T V^{-1} (\underline{S}_K - \underline{S}_j) > \underline{S}_K^T V^{-1} \underline{S}_K - \underline{S}_j^T V^{-1} \underline{S}_j$$
$$2\underline{N}_j^T V^{-1} (\underline{S}_K - \underline{S}_j) > \underline{S}_K^T V^{-1} \underline{S}_K - 2\underline{S}_j^T V^{-1} (\underline{S}_K - \underline{S}_j) - \underline{S}_j^T V^{-1} \underline{S}_j$$
$$2\underline{N}_j^T V^{-1} (\underline{S}_K - \underline{S}_j) > \underline{S}_K^T V^{-1} \underline{S}_K - 2\underline{S}_j^T V^{-1} \underline{S}_K + \underline{S}_j^T V^{-1} \underline{S}_j \tag{28}$$

Denote the left side of equation 28 by the random variable Y_K and the right side by the nonrandom variable Z_K. The mean of Y_K is zero and its variance is:

$$\mathrm{Var}\left\{Y_K\right\} = E\left[Y_K^2\right] \tag{29}$$
$$= 2\, E\left[\left(\sum_{i=1}^{M} \frac{n_{ij}\,(s_{iK} - s_{ij})}{V_i}\right)^2\right] \tag{30}$$
$$= 2\, E\left[\sum_{i=1}^{M} \frac{n_{ij}(s_{iK}-s_{ij})}{V_i}\left(\sum_{l=1}^{M} \frac{n_{lj}(s_{lK}-s_{lj})}{V_l}\right)\right] \tag{31}$$

Because the noise components are uncorrelated, equation 31 reduces to:

$$\mathrm{Var}\left\{Y_K\right\} = 2\sum_{i=1}^{M} (s_{iK} - s_{ij})^2 \tag{32}$$

So we write equation 28 in terms of the zero mean, unit variance random variable:

$$a_K > \frac{Z_K}{2\left|\underline{S}_K - \underline{S}_j\right|^2} = Z_{K1} \tag{33}$$

where $|\underline{S}_K - \underline{S}_j|$ is the Euclidean norm between the phonets of the pair of classes A_K and A_j, the phonet being approximated by the mean descriptor. The random variable a_K is distributed guassian with zero-mean and unit variance. So its density is:

$$P_a (a_K) = \frac{1}{\sqrt{2\pi}} \exp \left\{-\tfrac{1}{2}\, a_K^2\right\} \tag{34}$$

Then:

$$P\left\{E_K \middle| \underline{S}_j\right\} = \frac{1}{\sqrt{2\pi}} \int_{Z_{K1}}^{\infty} \exp \left\{-\tfrac{1}{2}\, a_K^2\right\} da_K \tag{35}$$

$$= Q(Z_{K1}) \tag{36}$$

The error probability conditioned on $\underline{S}_j$ is, from equation 24,

$$P\left\{E \middle| \underline{S}_j\right\} < \sum_{\substack{K=1 \\ K\neq j}}^{L} Q(Z_{K1}) \tag{37}$$

To bound the overall average error probability, equation 37 is averaged over all classes, so:

$$P \left\{E\right\} < \sum_{1=1}^{L} \frac{1}{L} \sum_{\substack{K=1 \\ K\neq j}}^{L} Q(Z_{K1}) \tag{38}$$

This average can be further upper-bounded in terms of the minimum Z_{K1}. That is, denote:

$$Z_{min} = \underset{1}{\text{Min}} \left[\underset{K\neq 1}{\text{Min}}\ Z_{K1}\right]. \tag{39}$$

Then:

$$P \left\{E\right\} < (L - 1)\ Q\ (Z_{min}). \tag{40}$$

The Q-function can be approximated (Ref 15:258) by:

$$Q(\chi) \leq \frac{1}{\sqrt{2\pi}} \frac{2 \exp\left\{-\frac{1}{2}\chi^2\right\}}{\chi + \sqrt{\chi^2 + \gamma/\pi}} \qquad (41)$$

so:

$$P\left\{E\right\} < \frac{2\ (L-1) \exp\left\{-\frac{1}{2} z_{min}^2\right\}}{\sqrt{2\pi}\ \left(z_{min} + \sqrt{z_{min}^2 + \gamma/\pi}\right)} \qquad (42)$$

One might relate the probability of classification error to the maximum number of classes, M_u, and the outer supervisory parameter in the following way. One could consider the minimum value of Z_K, the right side of equation 28, to be the signal-to-noise ratio in detecting the closest two classes. The signal-to-noise ratio at the input to the detector could be measured. It would be:

$$SNR = \frac{\underset{i=j}{\text{Min}} \left| \underline{S}_i - \underline{S}_j \right|^2}{\text{Var}\left\{\underline{N}\right\}} \qquad (43)$$

For a specified $P\left\{E\right\}$ in the detector, then,

$$z_{min} = Q^{-1}\left(\frac{P\left\{E\right\}}{(M_u - 1)}\right) \qquad (44)$$

from equation 40 can be thought of as the minimum distance normalized to the input variance at the detector. Scaling this distance by the SNR at the detector input, one gets the outer supervisor parameter from:

$$\lambda_u^2\ z_{min} = SNR. \qquad (45)$$

Once the feature space is partitioned into classes of phonetic units, then those phonetic units can be recognized in speech. Our Acoustic Analyzer, Seelandt's Speech Sound Analysis Machine (Ref 1) or some other distance computer, can be used to accomplish the recognition. The constraint, of course, is that the phonetic units used to partition feature space must be the same as those generated from speech by the distance computer.

In this chapter, we have presented the results of this project. We presented the Acoustic Analyzer and illustrated the behavior of observations under a minimum distance decision criterion. Then Conclusions and Recommendations are presented in Chapters IV and V, respectively.

IV. CONCLUSIONS.

We have implemented Seelandt's Speech Sound Analysis Machine (SSAM) on the Eclipse S250 Array Processor. We were able to generate phonetic units, which we called observations and phonets, from speech files. We calculated distances between observations and phonets, and we were able to output the best phonet choices along with other parameters for input to Montgomery's word recognizer (Ref 10). We also provided a graphics capability for plotting phonetic units, distances, and time files.

We also reviewed the parameters Seelandt identified for further study (Ref 1) and provided a basis for choosing parameter values. With the values we chose, we observed that adjacent phonetic units in voiced vowel sounds resemble each other in both distance rules. We found that while adjacent phonetic units in speaker-absent background noise and in "r" sounds did not resemble each other, they did in "s" sounds. We conclude that there seems to be a sufficient tendency for phonetic units to cluster to warrant further work along these lines.

We also adapted an algorithm for self-supervised vowel recognition, which used the first three formants as features in the absence of noise to our problem of phonetic unit recognition. The algorithm is explained and we showed how the S250 Array Processor and our Acoustic Analyzer could be used to implement it.

RECOMMENDATIONS.

We were unable to accomplish much important work that
s to be done before a phonet detector built along the
s of our Acoustic Analyzer can be optimized. One major
rt is needed in implementing an algorithm for partition-
the feature space. The algorithm by Pal, et. al. (Ref
that we considered in this project is one of many
ods. While we feel it has merit; we found many others
also warrant careful consideration. The recent book
evijver and Kittler (Ref 9) discusses the topic in
il. The text by Fukunaga (Ref 13) also treats the
ect and presents algorithms which could be applied to
problem. An algorithm should be implemented.

Once a method for partitioning feature space is imple-
ed, the parameters Seelandt discussed and that were
ted in this project, should be investigated experiment-
. That is, their effect on recognition of phonetic
s chosen by a feature space partitioning algorithm should
etermined.

Several features need to be added to our Acoustic
yzer. For one, the capability to overlap FFT windows
izes other than just 128 points needs to be incorporated.
routine S64, which allows the necessary window sizes but
not permit window overlapping, can serve as a departure

point for a general purpose FFT routine that incorporates the overlap feature, additional window shapes, and more sophisticated spectrum shaping options.

Also needed is the capability for the distance routines, DSTA and DSTN, to do a cross-correlation over shifts of only a few components between observation and phonet spectrum. In this way, the number of phonet clusters in feature space may be reduced if several phonet clusters are only shifted versions of others. Dete tion could proceed conditionally on frequency shift, and then choices between the shifts could be made. But before this is implemented, it may be valuable to implement first a partitioning algorithm. Once the feature space is partitioned, the cluster characteristics may suggest whether spectral shifts may offer improvement, how far to shift, and how much improvement. Answers to these questions depend heavily on the way that feature space is partitioned.

BIBLIOGRAPHY

1. Seelandt, Karl G. Computer Analysis and Recognition of Phoneme Sounds in Connected Speech, MS Thesis GE/EE/81D-53, School of Engineering, Air Force Institute of Technology, Wright-Patterson AFB, Ohio, December 1981.

2. Felkey, Mark A. Automatic Recognition of Phonemes in Continuous Speech, MS Thesis GE/EE/80D-20, School of Engineering, Air Force Institute of Technology, Wright-Patterson AFB, Ohio, December 1980.

3. De Souza, Peter and Thomson, Peter J. "LPC Distance Measures and Statistical Tests with Particular Reference to the Likelihood Ratio," IEEE Transactions on Acoustics, Speech, and Signal Processing, Vol ASSP-30, No. 2, April 1982.

4. Kassam, Salem A. A Bibliography on Nonparametric Detection, AD A080187, prepared for the Air Force Office of Scientific Research, AFOSR-TR-80-0025, 1979.

5. Lainiotis, D. G. Adoptive Systems, AD 776015, prepared for the Air Force Office of Scientific Research, AFOSR-TR-74-0309, February 1974.

6. Pal, S. K., Datta, A. K., and Dutta Majumder, D. "A Self-Supervised Vowel Recognition System," Pattern Recognition, Pergamon Press Ltd, Great Britain, Vol 12, 27-34.

7. Nonparametric/Robust Methods and Non-Guassian Models in Communication Theory, Johns Hopkins University, AD-763407, prepared for the Office of Naval Research, 1971.

8. Rabiner, L. R. and Schafer, R. W. Digital Processing of Speech Signals, Prentice Hall, 1978.

9. Devijver, R. P. and Kittler, J. Pattern Recognition: A Statistical Approach, Prentice Hall, 1982.

10. Montgomery, Gerald. Isolated Word Recognition Using Fuzzy Set Theory, MS Thesis GE/EE/82D-74, School of Engineering, Air Force Institute of Technology, Wright-Patterson AFB, Ohio, December 1982.

11. Sivian, L. J. "Speech and its Power Measurement," the Bell System Technical Journal, Vol 8, pp. 646-661, 1929.

12. Kuligowski, T. J. An Automatic Speech Teaching Responder, MS Thesis, United States Naval Postgraduate School, AD 851813, June, 1968.

13. Fukunaga, K. Introduction to Statistical Pattern Recognition, Academic Press Inc., 1972.

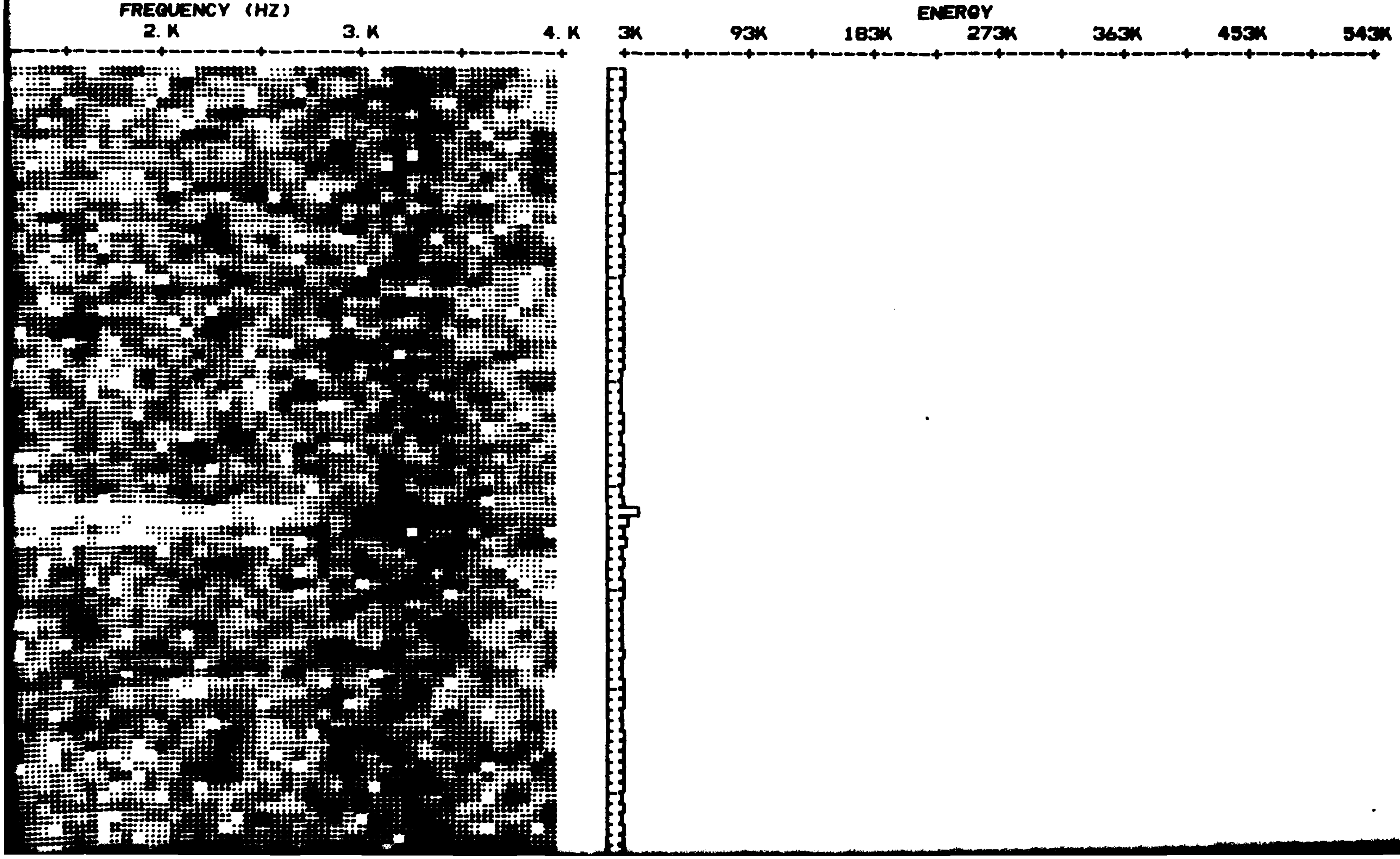
14. Wozencraft, J. M., and Jacobs, I. M. Principles of Communication Engineering, John Wiley and Sons, New York, 1965.

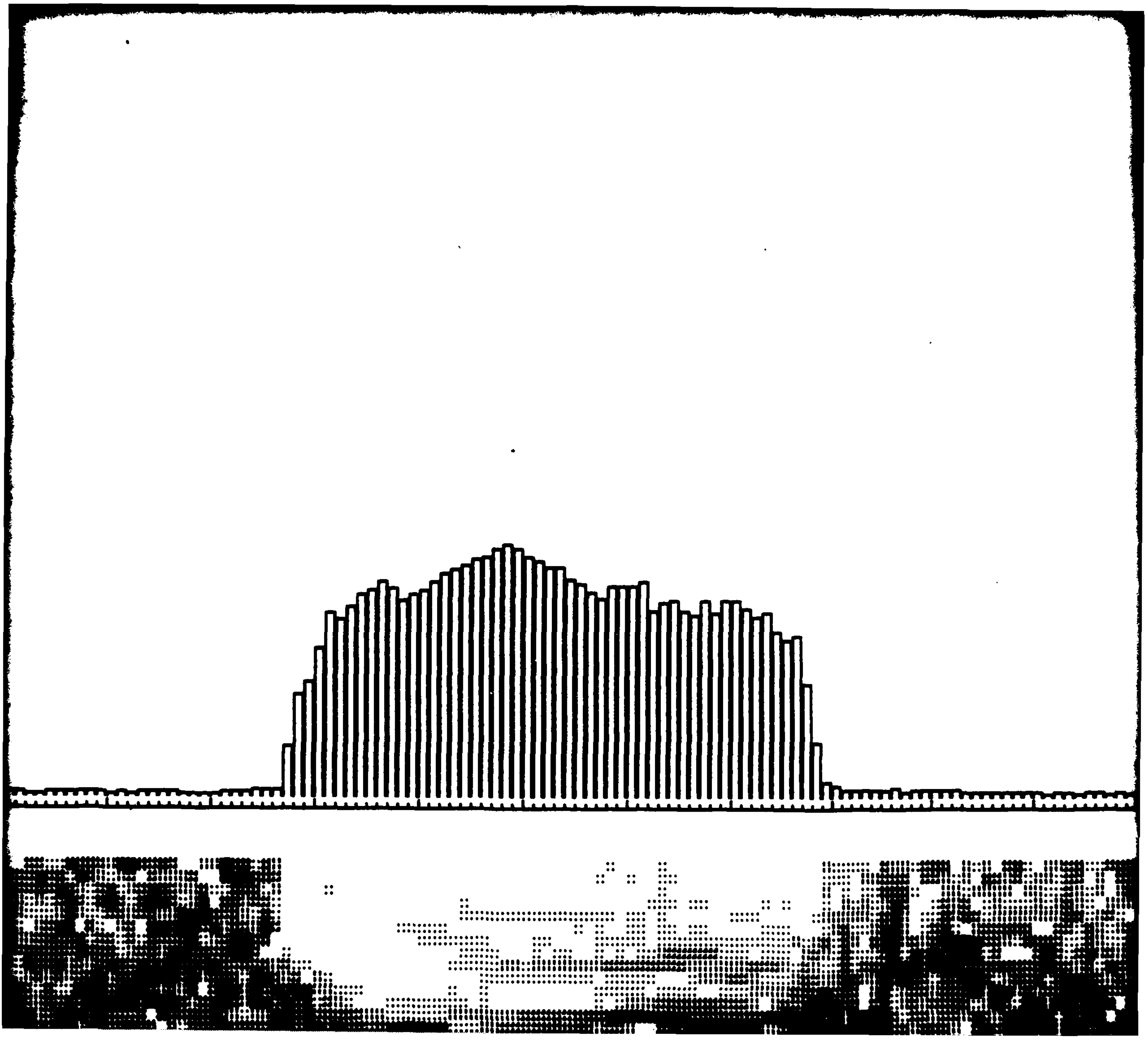
15. Melsa, J. L., and Cohn, D. L. Decision and Estimation Theory, McGraw-Hill, Inc., 1978.

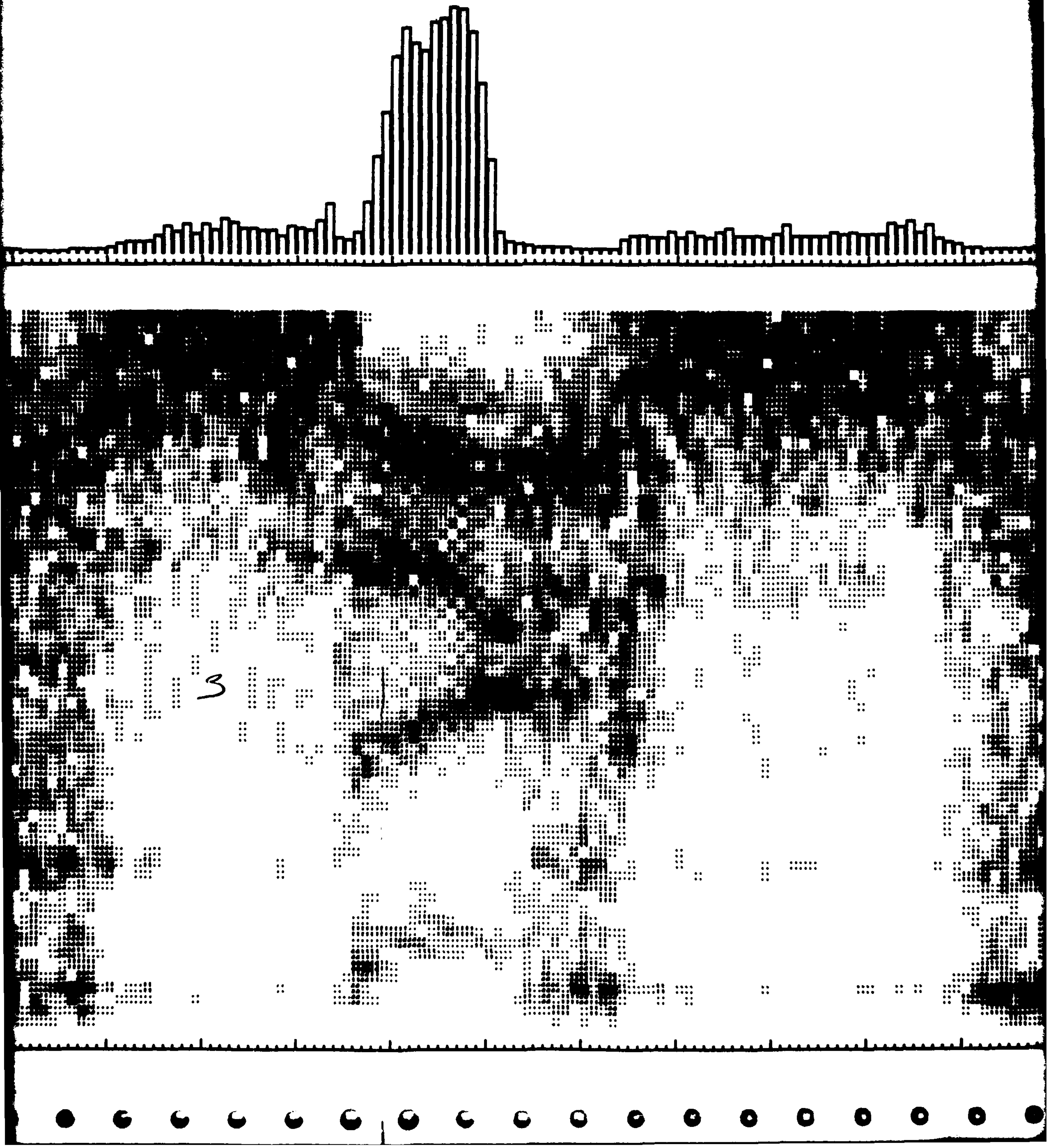
APPENDIX A

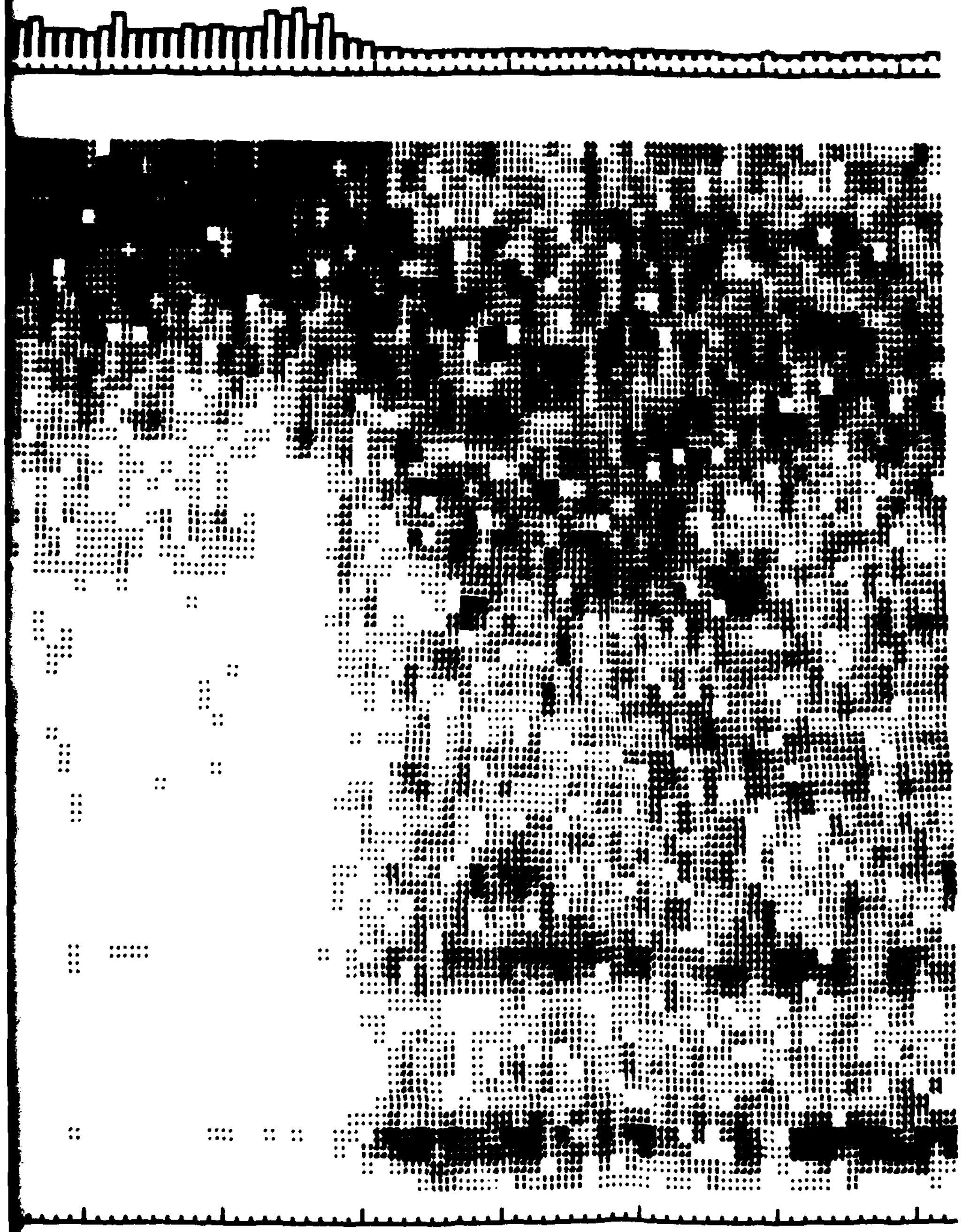
SPECTROGRAM OF FILE CT56.SP

```
ED ***
ENTENCE SPOKEN: FIVE SIX
IME: 11 49 21
  LAST TIME SLICE = 324

0% OVERLAP
CT ABOVE 300HZ.
T BELOW 300HZ.
```





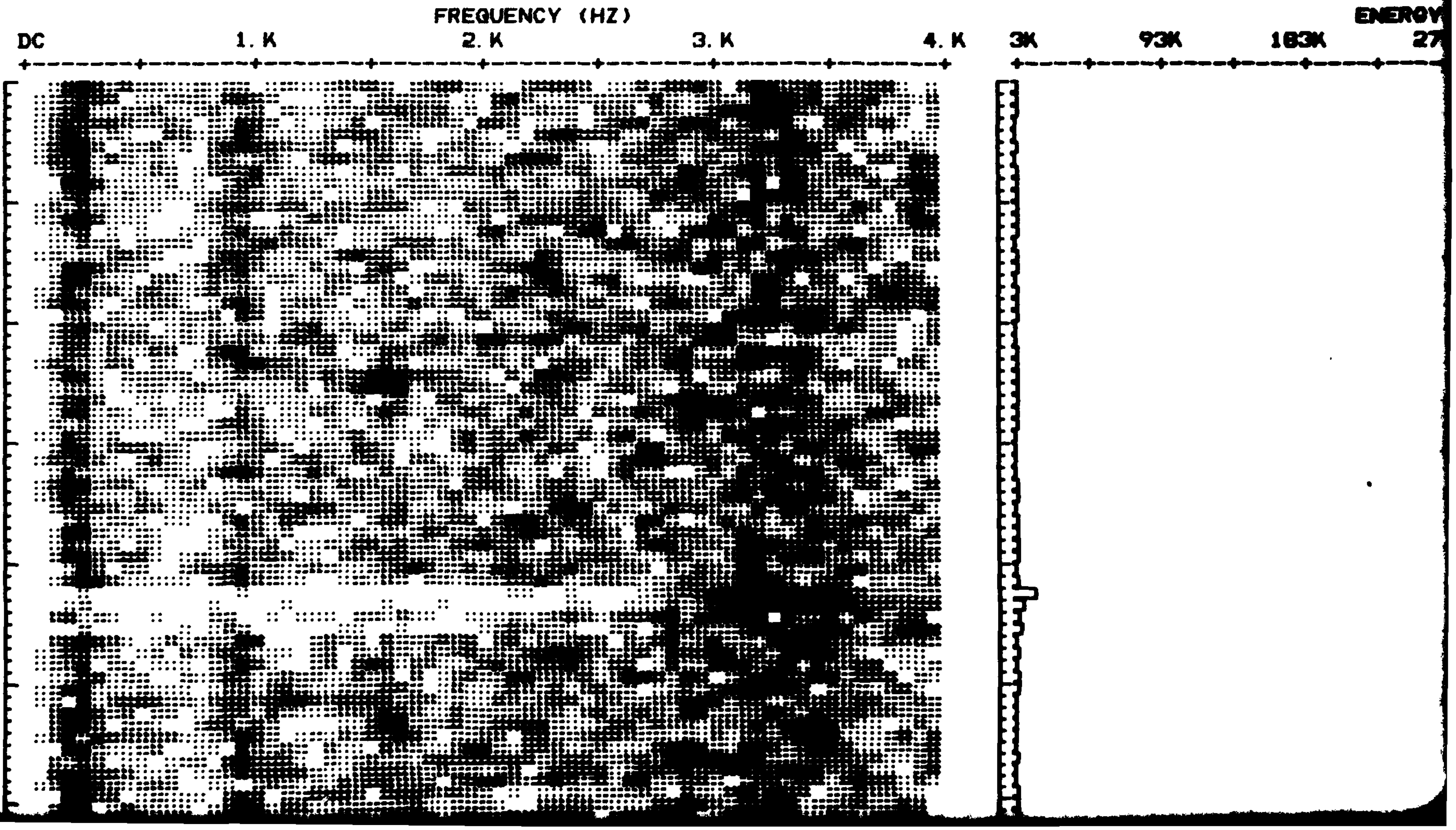

3

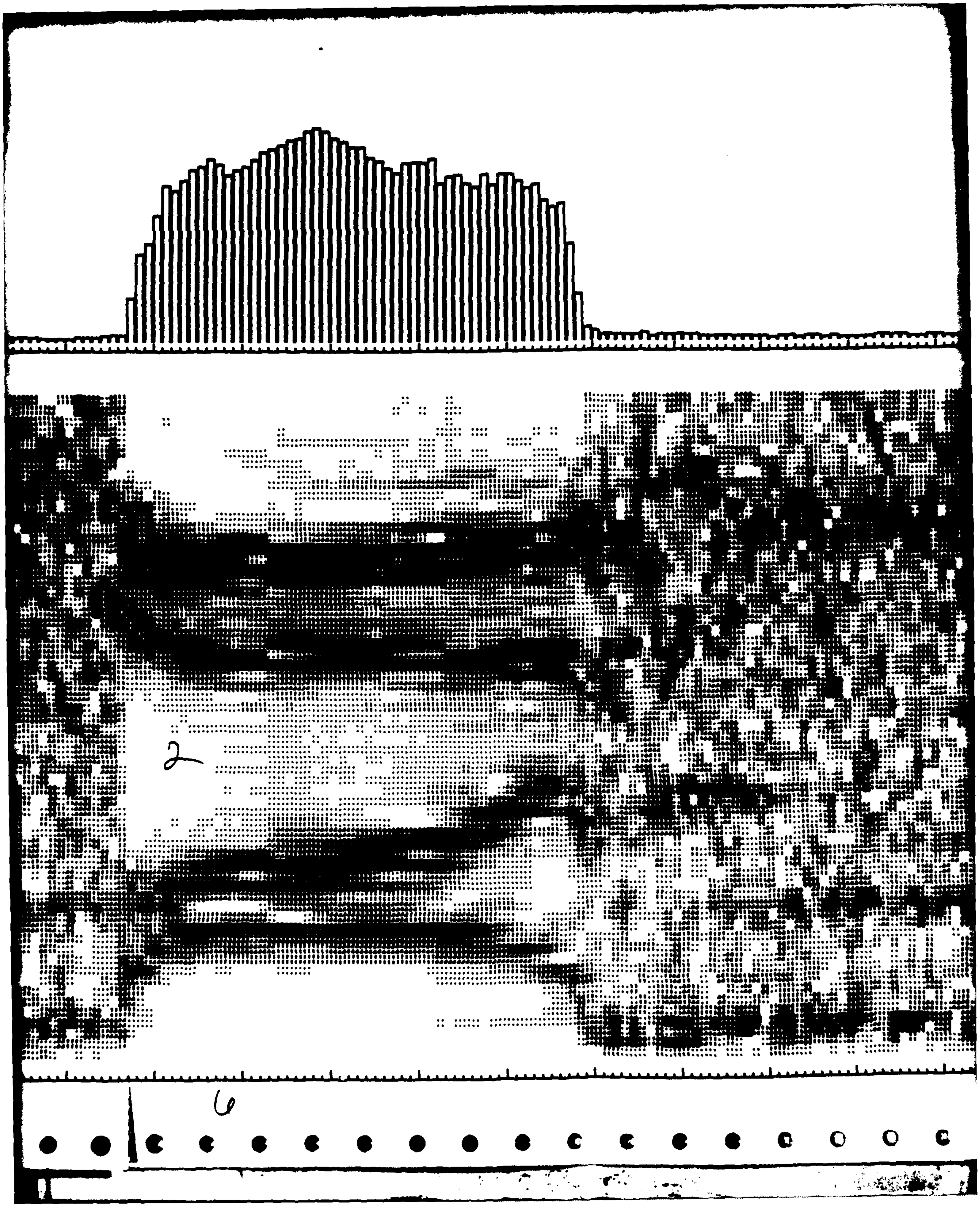


4

5

```
*** HAMMING WINDOW USED ***
FILE: CT56.OB        SENTENCE SPOKEN: FIVE SIX
DATE: 11 29 1982     TIME: 11 49 21
FIRST TIME SLICE = 1  LAST TIME SLICE = 324

DFT SIZE = 128       50% OVERLAP
PREEMPHASIS = 10DB/OCT ABOVE 500HZ.
DEEMPHASIS = 10DB/OCT BELOW 300HZ.
```




2
6

APPENDIX B

SOFTWARE DOCUMENTATION

STRUCTURE OF THE ACOUSTIC ANALYZER:

The Acoustic Analyzer consists of one executive routine, 15 subroutines, and calls to two library files. The executive routine, DRVR.SV, and subroutines CHKO, FCTR, GFDB, along with subroutines from the library files, are loaded into one save file, DRVR.SV. An overlay node is created in the save file into which segments of the overlay file, DRVR.OL, are loaded when called. The macro file which calls the loader utility is included in this documentation.

Figure 71 illustrates the software structure. As called, routines DRSQ, DSTN, DSTA, or PLTO are loaded into the executive routine at the overlay node, along with their supporting subroutines. The other routines: CHKO, FCTR, GFDB, and the libraries are accessible to the contents of the overlay node.

Below is a list of each routine in the Acoustic Analyzer, along with a brief statement of its purpose:

(1) DRVR is the executive. It allows the operator to select either the spectral option, either of the two distance options, or the plot options.

(2) DRSQ calls either S128 or S64 to compute the Fast Fourier Transform (FFT) of an input file.

(3) S128 computes FFTs of an input file using a 128 point window which overlaps adjacent windows by 64 points.

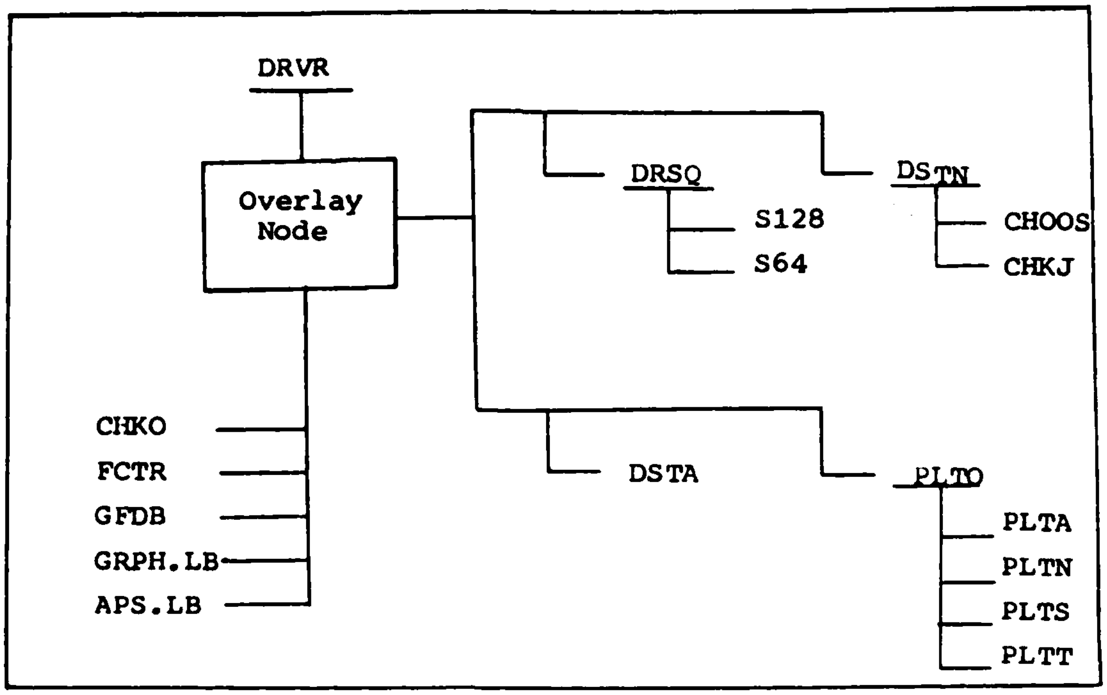


FIGURE 71. Structure of the Acoustic Analyzer. Options available from DRVR are program segments resident on disk and loaded into the overlay node on call.

(4) S64 computes FFTs of an input file using a nonoverlapped window of any length which is an integral multiple of two in the range 16-1024 points, inclusive.

(5) DSTA computes distances between observations and phonets, and stores the symmetric distance matrix on disk.

(6) DSTN computes the distance matrix that DSTA computes, but chooses a selected number of best template matches for output to disk.

(7) CHOOS is the subroutine that DSTN calls to find best matches between observations and phonets.

(8) CHKJ is called by DSTN to output data when distance-file elements are accumulated.

(9) PLTO allows the operator to call options h either display spectrum, display distances output STA, display distances output by DSTN, or display the ents of integer disk files.

(10) PLTA displays distance files computed by .

(11) PLTS displays spectral files computed by .

(12) PLTN displays distance files computed by .

(13) PLTT displays any segment of 512 integers ewer from a disk file.

(14) CHKO is called by DSTA and DSTN to type a ion statement to screen if a file unit number is in use.

(15) FCTR is called by GFDB to factor an integer a linear combination of four terms.

(16) GFDB is called by PLTA, PLTS, PLTN, and PLTT ompute the location of a specified data point in a file.

FILE:	LDDRVR.MC
LANGUAGE:	Command Line Interpreter
DATE:	September 21, 1982
AUTHOR:	D. Martin
SUBJECT:	Acoustic Analysis
CALLING SEQUENCE:	LDDRVR
DATE OF LAST REVISION:	September 21, 1982

PURPOSE:

This macro file loads program DRVR.SV. An overlay node is established and an overlay file, DRVR.OL, and load map, DRVR.LM, are created.

DESCRIPTION:

Location: DP4:BRATCHET

Size:	LDDRVR.MC	155 bytes
	DRVR.SV	51200 bytes
	DRVR.FR	2613 bytes
	DRVR.RB	3054 bytes
	DRVR.OL	40960 bytes
	DRVR.LM	9491 bytes

PROGRAM USE:

This macro uses the libraries GRPH.LB and APS.LB; a link to these are required. It loads the main program DRVR and the subroutines DRSQ, S128, S64, DSTA, DSTN, PLTA, PLTO, PLTS, PLTT, PLTN, CHKO, CHKJ, CHOOS, FCTR, and GFDB.

```
DELETE/V DRVR.LM
RLDR/C/R/E/P 2000/N DRVR [DRSQ S128 S64,DSTA,DSTN CHOOS CHKJ,^
PLTO PLTA PLTS PLTN PLTT] CHKO FCTR GFDB ^
DRVR.LM/L GRPH.LB APS.LB @FLIB@
```

FILE:	DRVR
LANGUAGE:	FORTRAN 5
DATE:	September 21, 1982
AUTHOR:	D. Martin
SUBJECT:	Acoustic Analysis
CALLING SEQUENCE:	DRVR
DATE OF LAST REVISION:	September 21, 1982

PURPOSE:

This program drives the acoustic analysis package developed by Captain Dan Martin for his thesis. When this program is executed, a menu is displayed from which the operator chooses an analysis option: to get spectrum, to get distances, to get best distances, or to plot results.

DESCRIPTION:

Location: DP4:BRATCHET

Size:	DRVR.SV	51200 bytes
	DRVR.FR	2613 bytes
	DRVR.RB	3054 bytes
	DRVR.OL	40960 bytes
	DRVR.LM	9491 bytes

PROGRAM USE:

CAUTION. The Array Processor must be properly initialized prior to any attempt to run this routine. A macro file, LDAPS.MC, in DP4:BRATCHET will accomplish this initialization. Also, at least 14 blocks of extended memory must be allocated to the ground on which this routine is run. Further, DRVR.OL, GRPH.LB, and APS.LB need be accessible to the directory in which this routine is run.

This program is not called by any routine and calls routines DRSQ, DSTA, DSTN, and PLTO. When executed, this program presents a menu from which the operator selects one of the following options:

(1) To compute observation spectrum from a speech file.

(2) To compute distances between observations and phonets. Phonets are observations set aside to represent recognizable units of speech.

(3) To compute distances between observations and phonets as in the previous option, but to choose a selected number of best matches based on those distances.

(4) To plot files on the Tektronix 4010-1 Graphics Terminal.

LIST OF VARIABLES:

VARIABLE	TYPE	PURPOSE
XMEM	Integer	Number of available extended memory blocks.
INITMEM	Integer	Number of extended memory blocks over and above those used in the logical address space window.
SFREQ	Real	Sampling rate for DRSQ.
DTARAY	Real Array	Vector in array processor memory.
SKIP	Integer	Switch to select options.

__SWITCH SETTINGS__:

SWITCH		SETTING
SKIP	=	1 to compute observations
	=	2 to compute distances
	=	3 to select best matches
	=	4 to plot files
	=	5 to stop

The remainder of this section is devoted to a discussion of: (1) our use of the S250 Array Processor, (2) extended memory management, and (3) modification of the Acoustic Analyzer (AA).

The Eclipse S250 Array Processor combines an Eclipse S250 CPU with a floating-point Array Processor. This allows high-speed computation on real and complex vectors. We used FORTRAN 5 as the application programming language and Data General's Array Processor Software (APS) running under Data General's mapped RDOS operating system. The AP/S250 contains both an independent pipelined floating-point multiplier and add/subtract compare units that operate simultaneously. Several multiply or add/subtract operations overlap during given time periods.

AP memory consists of an AP-mounted, 64 bit by 1K (8kByte), 20MSEC bipolar RAM. This memory is dual-ported which allowed us to address it by two methods:

(1) We addressed APM as part of main memory in which case we mapped APM into our 32K word logical

address space. In this way, data transfer between APM and logical address space where unnecessary.

(2) We addressed APM as local memory by using AP instructions.

We used the two categories of APS routines: support routines and interface routines. With the support routines, we handled the following tasks:

(1) AP initialization.

(2) Setting up control blocks and default values.

(3) AP memory management.

(4) Miscellaneous utility tasks.

With the interface routines, we performed specific Array Processing operations, such as to multiply two arrays, for example. We used first level interface routines, referred to as V-routines. Before executing a V-routine, we defined all data and control words required by the APS for that V-routine. This setup is accomplished by calling support routines as specified in the V-routine description. Besides control block words, we used a number of special APS parameters to make our application program independent of actual main memory addresses or APM offsets.

We managed memory in our Acoustic Analyzer (AA) using memory mapping and static memory allocation. We chose to declare a nine-memory block window in logical address space into which we would map APM and other arrays which we used as ports to extended memory. These port arrays: IDSP, IDOB in DRSQ, and DIST, FON, and OBS in DSTA and DSTN, were mapped through extended memory to access data with a minimum of disk accesses. This scheme is illustrated in Figure 7 for DSTA and DSTN. In extended memory, the first four memory blocks are dedicated to the AP; this fact cannot be altered by the application program. Our window in logical address space consists of the four real arrays: WKOBS, WKFON, OBS, and FON; and the one integer array DIST. Each array is 1024 elements long. Array Processor memory is mapped into WKOBS and WKFON. Arrays OBS, FON, and DIST are mapped through extended memory to access observation spectrum, phonet spectrum, or to store distances. From these arrays, data are parcelled to WKOBS and WKFON for processing. The scheme for DRSQ is the same except that the entire window is not used.

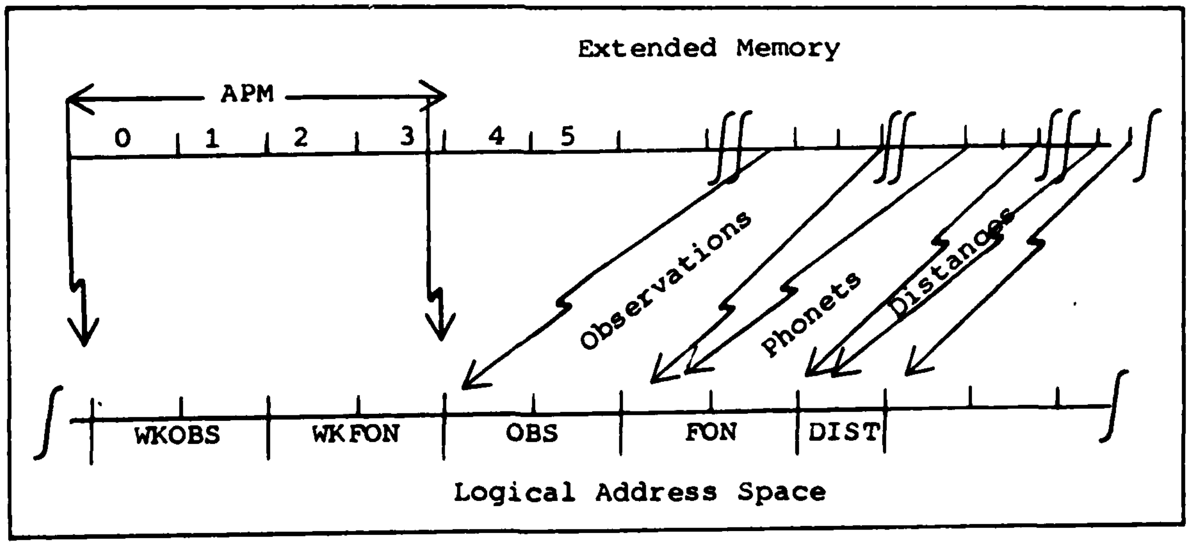


FIGURE 7. Memory Map for DSTA and DSTN

We have modified the Acoustic Analyzer for use by two students also doing their projects in the Speech Laboratory. We allowed them to run the spectral option and the distance option non-interactively by setting the switches in copies of the object code to their specifications. They now have access to the original interactive package with graphics capability as well as a tailored, non-interactive package capable of being run by MACRO files under RDOS. They use the output of the tailored Acoustic Analyzer as input to Montgomery's word recognition machine (Ref 10) which is also implemented on the Eclipse S/250 in the Speech Laboratory.

<u>RELATED PROGRAMS</u>:

PLTA, PLTS, PLTN, PLTO, PLTT, DRSQ, S128, S64, DSTA, DSTN, CHKJ, CHKO, CHOOS, F[illegible]TR, GFDB, GFPH2.

```
C****** ROUTINE DRVR.FR   NOTE: THIS ROUTINE IS FOR FORTRAN 5 ! !
C       THIS ROUTINE DRVES DRSQ,S128,S64,DSTA,DSTN,CHOOS
C       CHKJ,PLTO,PLTA,PLTS,PLTN,PLTT,CHKO,FCTR,CFDB.
C       BY:  CAPT DAN MARTIN
C       DATE:  9/27/82
C       SUBJ:  ACOUTSIC ANALYSER
C       THIS ROUTINE DRIVES THE ACOUSTIC ANALYSIS PACKAGE
C       I DEVELOPED.  THE OPERATOR IS PROMPED TO SPECIFY:
C               1)  AMOUNT OF EXTENDED MEMORY TO USE,
C               2)  TASK TO DO
C                       *GET SPECTRUM
C                       *GET DISTANCES
C                       *GET N-BEST DISTANCES
C                       *GET PLOTS
C                       *STOP
C       FOR DETAILS, SEE THE USERS MANUAL OR MY THESIS.


        INCLUDE "ARRAYP:F5APS.FR"
        EXTERNAL ODRSQ,ODSTA,ODSTN,OPLTO
        PARAMETER LM=512
        PARAMETER NDP=1024
        PARAMETER HLM=LM/2
        REAL DTARAY,B,PREM,WIND,SFREQ,IDOB
        INTEGER FILE1,IOPTN,IPRE,IDB,IHAM,FLEN,SKIP,INITMEM
        INTEGER ISTART,ILAST,COUNT,FILE2,SMOOTH,WRTD,TEST,FLAG
        INTEGER NBLKS,IFRQ(10),IAMP(10),XMEM,WRTDA,CB1,DUMMY
        INTEGER CNTBR,IDSP,SXMEM,CSPCT,IVAL
        COMMON /APM/ DTARAY(NDP),B(NDP)
        COMMON / VALS / IDSP(NDP),IDOB(NDP),IHAM,IPRE,WIND(LM)
        COMMON / VALS / PREM(LM),INITMEM,SXMEM,DUMMY(306)
        COMMON / VALT / SKIP,COUNT,SMOOTH,WRTD,NBLKS,XMEM,SFREQ,FLEN
        COMMON / VALT / FILE1(13),FILE2(13),FLAG,CNTBR,ISXMEM
        COMMON / VALT / JAOUT,LASTCD
        COMMON / VALV / CB1(0:CBMAX)
        COMMON / VALU / IVAL(2058)


30      CALL VMEM(XMEM,IER)
        INITMEM=XMEM-9
        XMEM=XMEM-1
35      TYPE"*YOU HAVE",XMEM,"  1K BLOCKS OF EXT MEMORY ACCESSABLE"
40      ACCEPT"**HOW MANY EXTENDED MEMORY 1K BLOCKS WILL YOU USE?  ",IXMEM
        IF(IXMEM.LE.XMEM)GOTO 45
        TYPE"YOU DON'T HAVE THAT MUCH MEMORY AVAILABLE."
        GOTO 35
45      XMEM=IXMEM-10
        IF(XMEM.GT.0)GOTO 50
        IXMEM=1-XMEM
        TYPE"***NEED SPECIFY AT LEAST",IXMEM,"  MORE BLOCKS!"
        GOTO 30
50      XMEM=IXMEM-9            ;LOSE 9 BLKS TO THE WINDOW
        ISXMEM=XMEM
        SFREQ=8000.0
        CALL RESET
        CALL OVOPN(20,"DRVR.OL",IER)
        CALL CHECK(IER)

C****** INITIALIZE ARRAY PROCESSOR AND MEMORY MAP
        CALL APINIT(NIL,DTARAY,9,INITMEM,IER)
        CALL APMAP(DTARAY,0,4,IER)

80      TYPE"**WHAT WILL YOU HAVE ME DO?"
        TYPE"****CHOOSE OPTION****"
        TYPE"*"
        TYPE"1 TO GET SPECTRUM."
        TYPE"2 TO GET DISTANCES."
        TYPE"3 TO GET N-BEST DISTANCES."
        TYPE"4 TO GET PLOTS."
        TYPE"5 TO STOP."
        ACCEPT"**ENTER OPTION DESIRED:  ",SKIP

100     IF(SKIP.NE.1)GOTO 110
        CALL OVLOD(ODRSQ,-1,IER)
        CALL CHECK(IER)
        CALL DRSQ
        GOTO 80
110     IF(SKIP.NE.2)GOTO 120
        CALL OVLOD(ODSTA,-1,IER)
        CALL CHECK(IER)
        CALL DSTA
        GOTO 80
120     IF(SKIP.NE.3)GOTO 130
        CALL OVLOD(ODSTN,-1,IER)
        CALL CHECK(IER)
        CALL DSTN
        GOTO 80
130     IF(SKIP.NE.4)GOTO 140
        CALL OVLOD(OPLTO,-1,IER)
        CALL CHECK(IER)
        CALL PLTO
        GOTO 80
140     IF(SKIP.EQ.5)GOTO 1000
        TYPE"ERROR: COULDN'T FIND YOUR OPTION."
        TYPE"YOU SELECTED OPTION:  ",SKIP
        GOTO 80
1000    CALL RESET
        STOP
        END
```

FILE:	DRSQ
LANGUAGE:	FORTRAN 5
DATE:	September 21, 1982
AUTHOR:	D. Martin
SUBJECT:	Acoustic Analysis
CALLING SEQUENCE:	DRSQ
DATE OF LAST REVISION:	September 21, 1982

<u>PURPOSE</u>:

This program calls subroutines S128 and S64 to compute the Fast Fourier Transform (FFT) of an input file. It uses the Eclipse AP/S250 Array Processor. Although flexible, these routines were designed to transform speech files produced by the Cromemco analog-to-digital system.

<u>DESCRIPTION</u>:

Location: DP4:BRATCHET

Size:	DRSQ.FR	11400 bytes
	DRSQ.RB	15840 bytes

<u>PROGRAM USE</u>:

This program is called by DRVR and calls S128, S64, and CHKO. This is a flexible routine in that several options are available with regard to windowing of input files, emphasis and scaling of output files. A block diagram is included at Figure 72.

When this routine is called, it prompts the operator for the following values and options:

(1) To pre-emphasize high frequencies or not?

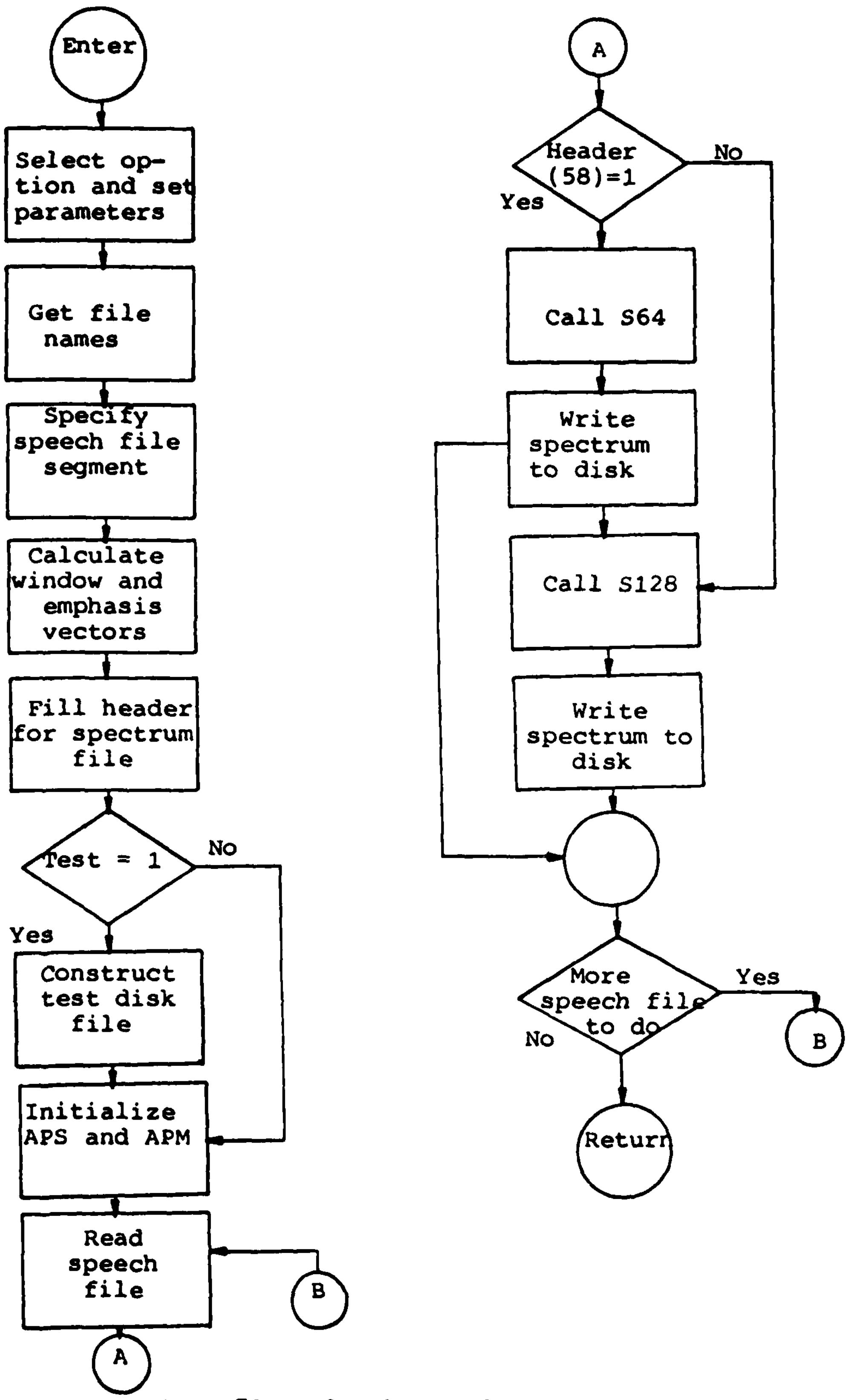


FIGURE 72. Flowchart of DRSQ

(2) If pre-emphasis is selected, at what rate in dB/Octave, and corner frequency?

(3) To de-emphasize low frequencies or not?

(4) If de-emphasis is selected, at what rate in dB/Octave, and corner frequency?

(5) What FFT window length? Any multiple of two between 16 and 1024, inclusive, can be selected.

(6) Choose Hamming or rectangular window. If a rectangular window is selected, no window vector is computed. If a Hamming window is selected, the window vector is calculated prior to the S128 or S64 call.

(7) If a window length of 128 points is selected, the program will prompt for a specification to overlap or not. If overlap is selected, the window will be moved along the input file in 64 point increments.

(8) To either normalize the observation energy to preset value or to divide each spectral component by the observation energy. The observation energy is the sum of the squares of each spectral component: one through N/2-1.

(9) Specify whether to write spectrum to disk only, to the terminal as well, or to the line printer as well.

(10) Specify if a test disk file is to be created and processed, or a speech file is to be read from disk and processed. DRSQ can generate a disk file with a signal

ı is a sum of up to ten tones at as many amplitudes. ›perator is prompted for these values if the test option ›lected.

(11) Specify speech file name.

(12) Specify a disk file name in which to store ›bservations. If this file does not exist, it will be :ed.

(13) Specify the portion of the speech file to ›ss. This is done by specifying the first and last blocks to process.

A header is prepared and written to disk block zero. › 1 contains the header assignments for the nontest ›n, Table 2 for the test option.

Extended memory blocks zero through eight are unavail- for use; they are used by the window declared in ›al address space. The speech file is read into all lable extended memory blocks and array IDSP is mapped ›gh extended memory accessing data. These data are ›lled from IDSP to Array Processor memory in FFT time ›s. After the FFT computation, the real spectrum is ›ferred through real array IDOB to extended memory. r extended memory contents have been replaced with rvations, its contents are written to disk.

HEADER ELEMENT	CONTENT
1-13	Observation file name.
14-26	Speech file name.
27	IPRE, pre-emphasis switch.
28	IDB, pre-emphasis rate (dB/Octave).
29	FREQ, frequency at which pre-emphasis starts.
30	FLEN, FFT window size.
31	IHAM, window switch.
32	Unused (set to zero).
33	TEST, test option switch.
34-54	Unused (set to zero).
55	COUNT, number of first time slice to do.
56	LTSTDO, number of last time slice to do.
57	HL, number of elements, per observation.
58	Overlapping switch.
59	Number of disk blocks in observation file.
60	IDPRE, de-emphasis switch.
61	DDP, de-emphasis rate (dB/Octave).
62	DFREQ, frequency at which de-emphasis ends.
63-256	Unused (set to zero).

TABLE 1. Header Assignments for Spectral File Computed by DRSQ Calling S128 or S64.

Test file header - but the header isn't written to disk.

HEADER ELEMENT	CONTENT
1-13	Test file name.
14-26	Observation file name.
27	Switch: 1 = pre-emphasize/0 = don't pre-emphasize.
28	Pre-emphasis slope.
29	Pre-emphasis corner frequency.
30	Number of time points-per-FFT.
31	Switch: 1 = Hamming window/0 = rectangular window.
32	Unused.
33	Switch: 1 = create test file/0 = don't create test file.
34	Number of tones in test file.
35-44	Tone frequencies (HZ).
45-54	Tone amplitudes.
55-57	Unused.
58	Switch: 1 = overlapping time slice/0 = nonoverlapping.
59-256	Unused.

TABLE 2. Header Assignments for Test Option.

LIST OF VARIABLES:

VARIABLE	TYPE	PURPOSE
XMEM	Integer	Number of extended memory blocks to use.
IOPTION	Integer	Switch to set parameter options.
IPRE	Integer	Pre-emphasis switch.
IDB	Integer	dB per Octave pre-emphasis.
FREQ	Integer	Starting pre-emphasis frequency.
IDPRE	Integer	De-emphasis switch.
DDB	Integer	dB per Octave de-emphasis.
DFREQ	Integer	Ending de-emphasis frequency.
FLEN	Integer	Number of points in FFT window.
IHAM	Integer	Switch to choose window.
HEADER	Integer Array	Holds header values.
NORM	Integer	Switch to select energy normalization.
WRTD	Integer	Switch to select output option.
TEST	Integer	Switch to select input option.
FILE1	Integer Array	To hold input file name.
FILE2	Integer Array	To hold output file name.
ISTART	Integer	First input disk block.
ILAST	Integer	Last input disk block.
LTSTDO	Integer	Last time slice to do.

VARIABLE	TYPE	PURPOSE
COUNT	Integer	First time slice to do.
WIND	Real Array	Holds window values.
PREM	Real Array	Holds emphasis values.
FREQ1	Integer	First component to pre-emphasize.
DFREQ	Integer	Last component to de-emphasize.
IDSP	Integer Array	Holds time file.
DTARAY	Real Array	First Array Processor Memory (APM) vector.
COUNTDOWN	Integer	Counts time slices done.
WRTDA	Integer	Output option switch.
NBLKS	Integer	Number of disk blocks to do.
LASTCD	Integer	Counts time slices to do.

SWITCH SETTINGS:

SWITCH		SETTING
IDPRE	=	1 to de-emphasize
	=	0 to not de-emphasize
IHAM	=	1 for Hamming window
	=	0 for rectangular window
HEADER(58)	=	1 to overlap time slices
	=	0 to not overlap
NORM	=	1 to normalize energy in time slice
	=	0 to divide spectrum by time slice energy
WRTD	=	0 to write output to disk only
	=	10 to write spectrum to terminal as well as disk
	=	12 to write spectrum to line printer as well as disk

SWITCH		SETTING
TEST	=	1 to create an integer disk file of tones to simulate speech
	=	0 to process speech
IOPTION	=	0 for ability to set parameters
	=	20-27 are option choices
IPRE	=	1 to pre-emphasize
	=	0 to not pre-emphasize
SKIP	=	0 to not execute APS and APM initialization
	=	1 never

RELATED PROGRAMS:

PLTA, PLTN, PLTS, PLTO, DSTA, DSTN, CHOOS, CHKJ, CHKO, S128, S64, GFDB, GRPH2, FCTR, PLTT, DRVR.

```
C****** SUBROUTINE DRSQ NOTE:  THIS SUBROUTINE IS FOR FORTRAN 5 !!
C       THIS SUBROUTINE DRIVES S128, S64.  IT IS CALLED BY DRVR.SV.
C       BY:  CAPT DAN MARTIN
C       DATE:  9/21/82
C       SUBJ:  ACOUSTIC ANALYSIS
C       THIS ROUTINE SETS UP NECESSARY FILES AND SWITCHES FOR
C       SUBROUTINES S128 AND S64, WHICH COMPUTE FOURIER TRANSFORMS
C       OF SPEECH FILES.  THE OPERATOR IS PROMPTED FOR NECESSARY
C       INFORMATION.
C       DRSQ DOES THE FOLLOWING:
C               1)  GETS PARAMETERS AND OPTIONS
C               2)  GETS SPEECH FILE
C               3)  GETS OBSERVATION (SPECTRAL) FILE
C               4)  GETS SPEECH FILE SEGMENT, SPECIFIED AS
C               1ST AND LAST DISK BLOCKS
C               5)  COMPUTES WINDOW VECTOR
C               6)  COMPUTES EMPHASIS VECTOR
C               7)  CONSTRUCTS SPECTRAL FILE HEADER
C               8)  CONSTRUCTS TEST SPEECH FILE IF THAT OPTION
C               WAS SELECTED
C               9)  SETS UP WINDOW AND MEMORY MAP FOR
C               ARRAY PROCESSOR
C               10)  READS SPEECH FILE INTO EXTENDED MEMORY
C               11)  CALLS EITHER S128 OR S64 TO GET SPECTRUM
C               12)  WRITES SPECTRUM FROM EXTENDED MEMORY TO
C               THE OBSERVATION FILE
C               13)  READS MORE SPEECH IF THERE IS MORE TO DO
C               14)  RETURNS TO DRVR
C       FOR MORE INFORMATION, SEE THE USERS MANUAL OR MY THESIS.

        SUBROUTINE DRSQ
        OVERLAY ODRSQ

        INCLUDE "ARRAYP:F5APS.FR".
        PARAMETER LM=512
        PARAMETER NDP=1024
        PARAMETER HLM=LM/2
        REAL DTARAY,B,PREM,WIND,SFREQ,IDOB
        INTEGER FILE1,IOPTN,IPRE,IDPRE,IDB,IHAM,FLEN,SKIP,HEADER(256),IDSP
        INTEGER ISTART,ILAST,COUNT,FILE2,SMOOTH,WRTD,TEST,DUMMY,FREQ
        INTEGER NBLKS,IFRQ(10),IAMP(10),XMEM,WRTDA,FLAG,CB1,CNTBR
        INTEGER INITMEM,SXMEM,GSPCT,LTSTDO,COUNTDOWN,NORM,DFREQ,DDB
        COMMON /APM/ DTARAY(NDP),B(NDP)
        COMMON / VALS / IDSP(NDP),IDOB(NDP),IHAM,IPRE,WIND(LM)
        COMMON / VALS / PREM(LM),INITMEM,SXMEM,DUMMY(304),COUNTDOWN
        COMMON / VALS / LTSTDO
        COMMON / VALT / SKIP,COUNT,SMOOTH,WRTD,NBLKS,XMEM,SFREQ,FLEN
        COMMON / VALT / FILE1(13),FILE2(13),FLAG,CNTBR,ISXMEM
        COMMON / VALT / JAOUT,LASTCD
        COMMON / VALV / CB1(0:CBMAX)
        COMMON / VALU / NORM

C******         Define variables used:
4       FLAG=0
```

```
        XMEM=ISXMEM
C****** GET SPECTRAL OPTIONS
5       TYPE"       Several sets of defaults exist for the"
        TYPE"   following parameters: "
        TYPE"       1) PREEMPHASIS (YES/NO)"
        TYPE"       2) PREEMPHASIS SLOPE (dB/octave)"
        TYPE"       3) PREEMPHASIS STARTING FREQUENCY"
        TYPE"       4) DEEMPHASIS (YES/NO)"
        TYPE"       5) DEEMPHASIS SLOPE"
        TYPE"       6) LAST FREQUENCY"
        TYPE"       7) FFT VECTOR LENGTH"
        TYPE"       8) WINDOW TYPE"
        TYPE"   They are: "

        TYPE"          OPTION NUMBER"

        TYPE"          #21      #22      #23      #24      #25      #26"
        TYPE"PREEMPH     Y        N        Y        N        Y       NO"
        TYPE"SLOPE      10       NA       10       NA       10       NA"
        TYPE"START     500       NA      500       NA      500       NA"
        TYPE"DEEMPH      Y        N        Y        N        N        N"
        TYPE"SLOPE      10       NA       10       NA       NA       NA"
        TYPE"LAST      300       NA      300       NA       NA       NA"
        TYPE"FFT LEN    64       64      128      128      128       64"
        TYPE"WINDOW     HM       HM       HM       HM       HM     RECT"
        TYPE"       If you choose to enter values for these parameters from"
        TYPE"   the keyboard, enter numeral zero.  If you choose a default"
        ACCEPT"    option, enter the option desired:  ",IOPTN

        IF(IOPTN.GT.20.AND.IOPTN.LT.27)GOTO 2050

        ACCEPT"PREEMPHASIZE HIGH FREQ ? (1=YES/0=NO): ",IPRE
        IF(IPRE.EQ.1)ACCEPT"ENTER dB/OCTAVE:  ",IDB
        IF(IPRE.EQ.1)ACCEPT"ENTER STARTING FREQ:   ",FREQ
        ACCEPT"DEEMPHASIZE LOW FREQ? (1=YES/0=NO):  ",IDPRE
        IF(IDPRE.EQ.1)ACCEPT"ENTER DB/OCTAVE:  ",DDB
        IF(IDPRE.EQ.1)ACCEPT"ENTER LAST FREQ:  ",DFREQ
        TYPE"ENTER FFT VECTOR LENGTH (SAMPLES/VECTOR).  LENGTH MUST"
        ACCEPT"BE A MULTIPLE OF 2 AND IN THE RANGE [16,1024]: ",FLEN
        TYPE"WHAT TYPE OF WINDOW IS DESIRED?"
        ACCEPT"HAMMING / RECTANGULAR (1/0): ",IHAM

        GOTO 50
2050    IF(IOPTN.EQ.21)GOTO 21
        IF(IOPTN.EQ.22)GOTO 22
        IF(IOPTN.EQ.23)GOTO 23
        IF(IOPTN.EQ.24)GOTO 24
        IF(IOPTN.EQ.25)GOTO 25
        IF(IOPTN.EQ.26)GOTO 26
        GOTO 50

21      IPRE=1                   ;preemphasize
```

```
        IDB=10                  ;dB/octave
        FREQ=500                ;start preemphasis here
        IDPRE=1
        DDB=10
        DFREQ=300
        FLEN=64
        IHAM=1
        GOTO 50
22      IPRE=0
        IDB=0
        FREQ=500
        IDPRE=0
        DDB=0
        DFREQ=0
        FLEN=64
        IHAM=1
        GOTO 50
23      IPRE=1
        IDB=10
        FREQ=500
        IDPRE=1
        DDB=10
        DFREQ=300
        FLEN=128
        IHAM=1
        GOTO 50
24      IPRE=0
        IDB=0
        FREQ=500
        IDPRE=0
        DDB=0
        DFREQ=0
        FLEN=128
        IHAM=1
        GOTO 50
25      IPRE=1
        IDB=10
        FREQ=500
        IDPRE=0
        DDB=0
        DFREQ=0
        FLEN=128
        IHAM=1
        GOTO 50
26      IPRE=0
        IDB=0
        FREQ=500
        IDPRE=0
        DDB=10
        DFREQ=300
        FLEN=64
        IHAM=0
        GOTO 50
```

```
C****** SET PARAMETERS
50      CONTINUE
        DO 415 J5=1,256
415     HEADER(J5)=0
        IF(FLEN.NE.128)GOTO 52
        TYPE"**ENTER '1' FOR OVERLAPPING"
        ACCEPT"**       '0' FOR NON-OVERLAPPING:  ",HEADER(58)
52      CONTINUE
        TYPE"***ENTER '1' TO NORMALIZE ENERGY IN FILE TO UNITY"
        ACCEPT"      OR '0' TO DIVIDE SPECTRUM BY FILE ENERGY:  ",NORM
        PARAMETER L=FLEN
        PARAMETER HL=L/2
C****** GET OUTPUT OPTION
        TYPE"*ENTER OUTPUT OPTION:  "
        TYPE"**0 TO WRITE SPECTRUM TO DISK (AND NO OTHER OUTPUT),"
        TYPE"**10 TO WRITE SPECTRUM TO SCREEN AS WELL AS DISK,"
        ACCEPT"**12 TO PRINT SPECTRUM AS WELL AS WRITE TO DISK. ",WRTD
C****** GET TEST OPTION
        TYPE"*GET TIME-FILE TYPE"
        TYPE"**ENTER A '1' TO CREATE A DISK FILE TO SIMULATE SPEECH:  "
        ACCEPT"**ENTER A '0' TO CRUNCH SPEECH:  ",TEST
        IF(TEST.EQ.0)GOTO 185
        ACCEPT"*ENTER TEST DISK FILENAME:  "
        READ(11,10)FILE1(1)
8       CALL OPEN(5,FILE1,2,IER)
        IF(IER.NE.1)TYPE"ERROR ON OPEN OF TEST DISK FILE, IER= ",IER
        CALL CHKO(IER)          ;SEE IF UNIT # IN USE
        IF(IER.NE.13)GOTO 9
        TYPE"FILE DOES NOT EXIST.  WILL CREATE IT FOR YOU."
        CALL CFILW(FILE1,2,IER1)
        IF(IER1.NE.1)TYPE"ERROR ON FILE CREATION. IER1= ",IER1
        CALL CHECK(IER1)
        GOTO 8
9       CONTINUE
        GOTO 190
C****** GET SPEECH FILENAME (IF NOT TEST)
185     ACCEPT "ENTER NAME OF SPEECHFILE: "
        READ(11,10)FILE1(1)
        CALL OPEN(5,FILE1,1,IER1)
        IF(IER1.NE.1)TYPE"ERROR ON OPEN OF SPEECHFILE, IER1= ",IER1
        CALL CHKO(IER1)         ;SEE IF UNIT # OPEN
190     CONTINUE
C****** GET OBSERVATION FILENAME
        ACCEPT"ENTER NAME OF OUTPUT DISK FILE:  "
        READ(11,10)FILE2(1)
10      FORMAT(S13)
12      CALL OPEN(4,FILE2,2,IER)
        IF(IER.NE.1)TYPE"ERROR ON OPEN OF OBSERVATION FILE, IER= ",IER
        CALL CHKO(IER)          ;SEE IF UNIT # OPEN
        IF(IER.NE.13)GOTO 15
        TYPE"FILE DOES NOT EXIST.  WILL CREATE IT FOR YOU."
        CALL CFILW(FILE2,2,IER1)
        IF(IER1.NE.1)TYPE"ERROR ON FILE CREATION. IER1= ",IER1
        CALL CHECK(IER1)
        GOTO 12
15      CONTINUE
C****** SPECIFY PORTION OF SPEECH FILE TO GET SPECTRUM OF, OR LENGTH
C       OF TEST INPUT FILE

        TYPE"**SPECIFY SPEECH FILE SEGMENT "
        TYPE"(MULTIPLE OF 4 DISK BLOCKS IF TEST FILE)"
        IF(TEST.EQ.0)GOTO 18
        TYPE"TEST TIME-DOMAIN FILE WILL START AT BLK #0"
        ISTART=0
        GOTO 19
18      ACCEPT"***ENTER FIRST BLOCK TO BE READ: ",ISTART
19      ACCEPT"***ENTER LAST BLOCK: ",ILAST
        IF(HEADER(58).EQ.0)LTSTDO=256*(1+ILAST)/L
                                                  ;LAST TIME-SLICE TO DO
        IF(HEADER(58).EQ.0)COUNT=ISTART*256/L+1
                                                  ;1ST TIME-SLICE TO DO
        IF(HEADER(58).NE.0)LTSTDO=256*(1+ILAST)/HL
                                                  ;LAST TIME-SLICE TO DO
        IF(HEADER(58).NE.0)COUNT=ISTART*256/HL+1
                                                  ;1ST TIME-SLICE TO DO

C****** COMPUTE WINDOW VECTOR
        TYPE"COMPUTING WINDOW VECTOR (IF REQUESTED)"
        IF(IHAM.NE.1)GOTO 210
        DO 200 J=1,L
        WIND(J)= 0.54 - (0.46*COS(6.2831853072*(J-1)/L))
200     CONTINUE

C****** CALCULATE EMPHASIS VECTOR
210     TYPE"CALCULATING EMPHASIS VECTOR (IF REQUESTED)"
        IF(IPRE.NE.1.AND.IDPRE.NE.1)GOTO 408
        DO 405 IN=1,HL
405     PREM(IN)=1.0
        IF(IPRE.NE.1)GOTO 295   ;TO NOT PREEMPHASIZE
        FREQ1=(FREQ/SFREQ)*L+1  ;FIRST FREQ PREEMPH
        DO 400 IN=FREQ1,HL
400     PREM(IN)=(IN/FREQ1)**(0.1660964*2*IDB)
295     CONTINUE

        IF(IDPRE.NE.1)GOTO 350  ;TO NOT DEEMPHASIZE
        FREQ1=(DFREQ/SFREQ)*L   ;LAST FREQ
        DO 300 IN=1,FREQ1
300     PREM(IN)=(IN/FREQ1)**(0.1660964*2*DDB)
350     CONTINUE
408     CONTINUE
C****** CONSTRUCT HEADER FOR OUTPUT FILE
        DO 410 J5=1,13
410     HEADER(J5)=FILE2(J5)
        DO 412 J5=14,26
        J4=J5-13
412     HEADER(J5)=FILE1(J4)
        HEADER(27)=IPRE
        HEADER(28)=IDB
        HEADER(29)=FREQ
        HEADER(30)=FLEN
        HEADER(31)=IHAN
        HEADER(33)=TEST
        HEADER(55)=COUNT
        HEADER(56)=LTSTDO
        HEADER(57)=HL
        HEADER(60)=IDPRE
        HEADER(61)=DDB
        HEADER(62)=DFREQ
        IF(IDPRE.EQ.1)IPRE=1
        WRTDA=WRTD
        IF(WRTD.EQ.0)WRTD=10
        WRITE(WRTD,950)HEADER(1)
        WRITE(WRTD,952)HEADER(14)
        WRITE(WRTD,954)
        WRITE(WRTD,956)(HEADER(J5),J5=27,31)
        WRITE(WRTD,955)
        WRITE(WRTD,956)(HEADER(J5),J5=60,62)
        WRITE(WRTD,958)HEADER(33)
416     IF(TEST.EQ.1)GOTO 430
420     GOTO 500
C****** FILL TEST DISK FILE
430     IS=1
        DO 432 JJ1=1,10
        IFRQ(JJ1)=0
432     IAMP(JJ1)=0
        ACCEPT"*ENTER 1ST  FREQ IN HZ: ",IFRQ(IS)
445     ACCEPT"*ENTER AMPLITUDE: ",IAMP(IS)
        ACCEPT"*****WANT ANOTHER TONE? (1-YES/0-NO): ",IT
        IF(IT.EQ.0)GOTO 455
        IS=IS+1
        IF(IS.GT.10)GOTO 455
        ACCEPT"*ENTER ANOTHER  FREQ IN HZ: ",IFRQ(IS)
        GOTO 445
455     IF(IS.GT.10)IS=10
        HEADER(34)=IS
        DO 456 J5=35,44
        J4=J5-34
456     HEADER(J5)=IFRQ(J4)
        DO 457 J5=45,54
        J4=J5-44
457     HEADER(J5)=IAMP(J4)
        WRITE(WRTD,962)HEADER(34)
        WRITE(WRTD,960)(HEADER(J5),J5=35,HEADER(34)+34)
        WRITE(WRTD,960)(HEADER(J5),J5=45,HEADER(34)+44)
459     DO 495 JI=0,ILAST,4
        DO 460 IT=1,NDP
460     IDSP(IT)=0.0
        DO 470 IQ=1,IS
        DO 450 IT=1,NDP
450     IDSP(IT)=IAMP(IQ)*COS(6.2831853072*IFRQ(IQ)*(IT-1)/SFREQ)+IDSP(IT)
470     CONTINUE
        CALL WRBLK(5,JI,IDSP,4,IER)
```

```
495     CONTINUE

500     CALL FGTIME(IHOURO,IMINO,ISECO)

        SKIP=0
C****** INITIALIZE APS AND APM MAP FOR ERDB
        IF(SKIP.EQ.0)GOTO 502
        SKIP=0
        CALL APINIT(NIL,DTARAY,9,INITMEM,IER)
        CALL APMAP(DTARAY,0,4,IER)
502     JI=1                              ;DISK BLOCK TO WRITE TO
        JIH=1                             ;DISK BLOCK TO WRITE TO
503     IF(HEADER(58).EQ.0)XMEM=XMEM*4    ;GET * AVAIL QTR BLKS EXT MEM
        IF(HEADER(58).NE.0)XMEM=XMEM*2
504     COUNTDOWN=HEADER(56)-HEADER(55)+1
        TYPE"COUNTDOWN=",COUNTDOWN
        IF(WRTDA.EQ.0)WRTD=0
505     NBLKS=ILAST-ISTART+1
        IF(NBLKS.LE.XMEM)GOTO 510
        NBLKS=XMEM               ;* QTR BLKS CAN DO
510     IF(HEADER(58).NE.0)GOTO 512
        COUNT=256*ISTART/L                ;INITIALIZE TIME SLICE COUNT
        CALL ERDB(5,ISTART,36,NBLKS,ICNT,IER)
        GOTO 530
512     COUNT=256*ISTART/HL
        CALL ERDB(5,ISTART,36+XMEM,NBLKS,ICNT,IER)
530     CONTINUE
        IF(IER.NE.9)GOTO 850
        TYPE"READ END OF FILE!  SUCCESSFULLY TRANSFERRED",ICNT,"  QTR BLOCKS."
        NBLKS=ICNT
        ILAST=ISTART+NBLKS-1
        TYPE"PROCEEDING WITH  ",NBLKS,"  QTR BLOCKS TRANSFERRED."
        IER=1
850     IF(IER.EQ.1)GOTO 860
        TYPE"ERROR ON ERDB,IER= ",IER
        GOTO 1000
860     ISTART=ISTART+NBLKS
2000    FORMAT(5G12.4)
        IF(FLEN.EQ.128.AND.HEADER(58).NE.0)GOTO 890
        LASTCD=COUNTDOWN-NBLKS*256/L
        CALL S64
        CALL EWRB(4,JIH,36,NBLKS,ICNT,IER)
        GOTO 904
890     LASTCD=COUNTDOWN-NBLKS*256/HL
        CALL S128
902     CALL EWRB(4,JI,36,NBLKS*2,ICNT,IER)
904     IF(IER.NE.1)TYPE"ERROR ON EWRB, IER= ",IER
        IF(COUNTDOWN.LE.0.AND.IER.EQ.9)NBLKS=ICNT
905     JI=JI+NBLKS*2
        JIH=JIH+NBLKS
        TYPE"***COUNTDOWN=",COUNTDOWN
        IF(COUNTDOWN.LE.0)GOTO 911
910     IF(ISTART.LE.ILAST)GOTO 505     ;MORE SPEECHFILE TO DO
```

```
911     CONTINUE
        IF(FLEN.EQ.128.AND.HEADER(58).NE.0)GOTO 912
        HEADER(59)=JIH          ;# QB'S IN FILE
        GOTO 915
912     HEADER(59)=JI
915     CALL WRBLK(4,0,HEADER,1,IER)            ;WRITE HEADER TO 1ST QBLK
        CALL RESET
        WRTD=WRTDA
        CALL FGTIME(IHOUR,IMIN,ISEC)
        ISECT=ISEC-ISEC0+60*(IMIN-IMIN0)+3600*(IHOUR-IHOUR0)
        TYPE"TIME TO GET SPECTRUM IS",ISECT,"  SECONDS"


950     FORMAT("****  THIS FILE'S NAME IS:    ",S13)
952     FORMAT("      TIME DOMAIN FILE NAME:  ",S13)
954     FORMAT(7X,"IPRE",6X,"IDB",6X,"FREQ",6X,"FLEN",6X,"IHAM")
955     FORMAT(7X,"IDPRE",7X,"DDB",6X,"DFREQ")
956     FORMAT(6I10)
958     FORMAT(" ---- TEST = ",I4"  ---")
960     FORMAT("  ",10I7)
962     FORMAT(2X,I4,"  TONES IN TEST FILE.  HERE ARE FREQS AND AMPLITUDES")
1000    CALL OVEXIT(ODRSQ,IER)
        CALL CHECK(IER)
        RETURN
        END
```

E:	S128 and S64
;UAGE:	FORTRAN 5
E:	September 21, 1982
HOR:	D. Martin
JECT:	Fourier Transform
LING SEQUENCE:	S128 or S64
E OF LAST REVISION:	September 21, 1982

POSE:

Subroutine S128, called by DRSQ, computes overlapping point Fast Fourier Transforms (FFTs) on an integer e residing in extended memory. Subroutine S64 does not rlap the input window and will do FFTs of window sizes ch are multiples of two in the range 16-1024, inclusive.

CRIPTION:

Location: DP4:BRATCHET

Size:	S128.FR	4866 bytes
	S128.RB	5378 bytes
	S64.FR	3910 bytes
	S64.RB	4276 bytes

3RAM USE:

These programs are called by DRSQ and call no sub-tines. They use the Array Processor and extended memory naximize the speed with which they calculate FFTs. These routines are similar and the flowchart, Figure 73, applies ooth. The two differences between them are: (1) the low size of S128 is fixed at 128 points while that of S64 selectable, (2) the window of S128 is moved along the

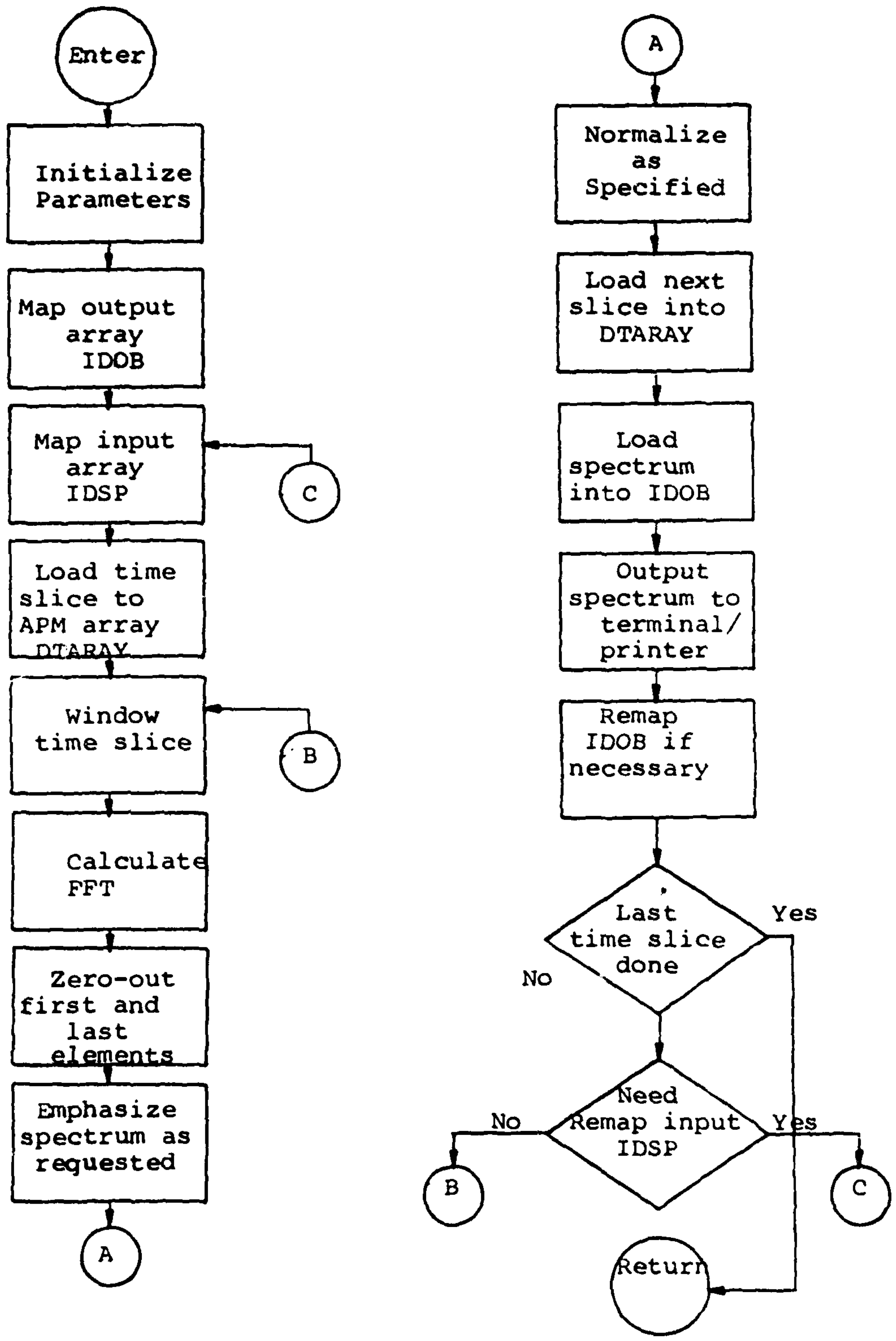


FIGURE 73. Flowchart of S128 and S64

input file in increments of 64 points at each FFT calculation while the window in S64 is moved along the input file in increments of its window size.

These differences require that extended memory be managed differently in the two routines. For S64, an input window of N integer points returns N/2 real components which replace the input time slice-by-time slice. The spectrum is stored on disk in contiguous units called observations. The first element is the observation energy, E, computed as the sum of the squares of spectral components one through N/2-1. The Array Processor FFT routine returns N/2+1 components; but prior to the energy computation, the zero and N/2 components are set to zero. Each observation, then, consists of an energy value as its first element, and spectral components one through N/2-1 in the remaining N/2-1 elements. The K^{th} element of the observation is the K^{th} spectral component of the N- point time slice.

The observations are written to disk starting at disk block number one. Each N- point time slice requires N words of memory. Taking advantage of this, S64 stores the spectrum back into the storage locations previously occupied by the window. In this sense, the FFTs are computed in-place.

But because S128 overlaps its window by 64 points each FFT calculation, the in-place scheme will not work. The scheme S128 uses has DRSQ read the input file into the upper half of available extended memory. Then as the input window is moved along the upper half, the output is stored in the lower half of extended memory, starting at the first available location. The input and output storage locations overlap only at the last locations at the top of extended memory.

Both routines transform an integer input file, such as that produced by the Cromemco analog-to-digital conversion system in the Speech Laboratory, into a real disk file. Disk block number zero is a header containing information which describes and identifies the file. This header is the same for both S128 and S64, and its assignments are listed in Table 3.

LIST OF VARIABLES:

VARIABLE	TYPE	PURPOSE
JAOUT	Integer	First element of output vector.
DUMMY(1)	Integer	Last memory block read from.
J6	Integer	First memory block read from.
J6S	Integer	Signals first call.
XMEM	Integer	Extended memory available for input.
DTARAY	Real Array	Array Processor memory.
IDSP	Integer Array	Input array mapped through extended memory.
WIND	Real Array	Contains window to be applied to input.

VARIABLE	TYPE	PURPOSE
PREM	Real Array	Contains emphasis vector applied to spectrum.
NORM	Integer	Switch to select energy normalization.
IPRE	Integer	Switch to select emphasis.
IHAM	Integer	Switch to select window type.
IDOB	Real Array	Output array mapped through extended memory.

SWITCH SETTINGS:

SWITCH		SETTING
IHAM	=	1 for Hamming window
	=	0 for rectangular window
IPRE	=	1 to emphasize spectrum
	=	0 to not
NORM	=	1 to normalize spectral components to observation energy
	=	0 to divide each spectral component by the observation energy

RELATED PROGRAMS:

PLTA, PLTS, PLTN, PLTT, PLTO, DRVR, DSTA, DSTN, DRSQ, CHKJ, CHKO, CHOOS, FCTR, GFDB, GRPH2.

HEADER ELEMENT	CONTENT
1-13	Observation file name.
14-26	Speech file name.
27	IPRE, pre-emphasis switch.
28	IDB, pre-emphasis rate (dB/Octave).
29	FREQ, frequency at which pre-emphasis starts.
30	FLEN, FFT window size.
31	IHAM, window switch.
32	Unused (set to zero).
33	TEST, test option switch.
34-54	Unused (set to zero).
55	COUNT, number of first time slice to do.
56	LTSTDO, number of last time slice to do.
57	HL, number of elements, per observation.
58	Overlapping switch.
59	Number of disk blocks in observation file.
60	IDPRE, de-emphasis switch.
61	DDP, de-emphasis rate (dB/Octave).
62	DFREQ, frequency at which de-emphasis ends.
63-256	Unused (set to zero).

TABLE 3. Header Assignments for Spectral File Computed by DRSQ Calling S128 or S64.

```
C****** SUBROUTINE S128 NOTE:  THIS ROUTINE IS FOR FORTRAN 5 !!
C       THIS SUBROUTINE CALLS ARRAY PROCESSOR ROUTINES TO CALCULATE 128
C       POINT FFT'S.  IT OVERLAPS THE TIME WINDOWS BY 64 POINTS.
C       BY:  CAPT DAN MARTIN
C       DATE:  9/21/82
C       SUBF:. ACOUSTIC ANALYSIS
C       THIS ROUTINE IS CALLED BY DRSQ WHICH SETS UP THE NECESSARY
C       FILE INPUT/OUTPUT FILES AND OPTIONS.  THE INPUT TO THIS ROUTINE
C       RESIDES IN EXTENDED MEMORY AND THE OUTPUT IS DEPOSITED THERE.
C       S128 DOES THE FOLLOWING:
C               1)  INITIALIZES PARAMETERS
C               2)  MAPS OUTPUT ARRAY IDOB
C               3)  MAPS INPUT ARRAY IDSP
C               4)  LOADS TIME SLICE INTO DTARAY
C               5)  WINDOWS TIME SLICE
C               6)  CALCULATES FFT
C               7)  ZEROS-OUT 1ST AND LAST ELEMENTS
C               8)  EMPHASIZES SPECTRUM
C               9)  NORMALIZES SPECTRUM
C               10)  LOADS SPECTRUM INTO IDOB
C               11)  REMAPS IDOB IF NECESSARY
C               12)  REMAPS IDSP IF NECESSARY
C       FOR MORE INFORMATION, SEE THE USERS MANUAL OR MY THESIS.
        SUBROUTINE S128
        INCLUDE "ARRAYP:F5APS.FR"     ; APS PARAMETER FILE

C******     Set up variables to be used.
        PARAMETER LM=64
        PARAMETER TLM=128
        PARAMETER NDP=1024
        PARAMETER MAXLM=512
        REAL DTARAY,B,PREM,WIND,SFREQ
        REAL SUME,FACTOR,IDOB
        INTEGER CB1,DUMMY,FLEN,FILE1,FILE2,FLAG,CNTBR,IDSP
        INTEGER IHAM,IPRE,SKIP,COUNT,SMOOTH,WRTD,NBLKS,XMEM
        INTEGER INITMEM,SXMEM,GSPCT,LTSTDO,COUNTDOWN,NORM

        COMMON /APM/ DTARAY(NDP),B(NDP)          ;AP MEMORY
        COMMON / VALS / IDSP(NDP),IDOB(NDP),IHAM,IPRE,WIND(MAXLM)
        COMMON / VALS / PREM(MAXLM),INITMEM,SXMEM,DUMMY(304),COUNTDOWN
        COMMON / VALS / LTSTDO
        COMMON / VALT / SKIP,COUNT,SMOOTH,WRTD,NBLKS,XMEM,SFREQ,FLEN
        COMMON / VALT / FILE1(13),FILE2(13),FLAG,CNTBR,ISXMEM
        COMMON / VALT / JAOUT,LASTCD
        COMMON / VALV / CB1(0:CBMAX)
        COMMON / VALU / NORM


C****** INITIALIZE APS AND APM MAP
        IF(SKIP.EQ.0)GOTO 250
        CALL APINIT(NIL,DTARAY,9,INITMEM,IER)
        CALL APMAP(DTARAY,0,4,IER)
250     CONTINUE
        JAOUT=1                            ;1ST ELEMENT OF OUTPUT VECTOR
        DUMMY(1)=9+(XMEM+NBLKS)/4                ;LAST MEM BLK READ FROM
        J6=XMEM/4+9                              ;1ST MEM BLK READ FROM
        J6S=J6                                   ;SIGNALS 1ST ENTRY
        J5=9
        CALL APMAP(IDOB,J5,-2,IER)               ;MAP OUTPUT ARRAY
260     CONTINUE
        CALL APMAP(IDSP,J6,-1,IER)               ;MAP INPUT ARRAY
C****** ITERATE TO LINE 700
C****** THE FIRST ELEMENT OF DATA DOESN'T RESIDE ON A 1K BOUNDARY,
C       NECESSARILLY, IN THE FIRST MAP.  TO ACCOUNT
C       FOR THAT POSSIBILITY:
        IF(J6.NE.J6S)J1=1                        ;GET 1ST ELEMENT OF IDSP
        IF(J6.EQ.J6S)J1=1+NDP*(FLOAT(XMEM)/4-IFIX(XMEM/4))
        J6=J6+1
C****** TRANSFER 128 PNTS FROM IDSP
280     DO 300 J2=J1,J1+TLM-1
        J3=J2-(J1-1)
300     DTARAY(J3)=IDSP(J2)



C****** WINDOW VECTOR DTARAY
305     IF(IHAM.NE.1)GOTO 310
        CALL APSETL(TLM,IER)
        CALL CBSET(CB1,CBAXR,B,CBAAMM,WIND,IER)
        CALL VLDR(CB1)                  ;LOAD WINDOW VECTOR
        CALL CBSET(CB1,CBAXR,DTARAY,CBAYR,B,CBAZR,DTARAY,IER)
        CALL VMRA(CB1)                  ;MULTIPLY VECTORS


  310   CONTINUE
C****** CALCULATE FFT OF VECTOR DTARAY

        CALL APSETL(LM,IER)

        CALL CBSET(CB1,CBAXC,DTARAY,CBCW,CWDFT,IER)
        CALL VFFTC(CB1)
        CALL VBRC(CB1)
        CALL VFFTR(CB1)
C****** COMPLEX DTARAY CONTAINS 1ST LM+1 SPECTRUM COEFFICIENTS
C       NEED ZERO-OUT 1ST ELEMENT, I.E., ZERO-HERTZ TERM
C       ALSO NEED ZERO-OUT 65TH ELEMENT, I.E., 4KHZ TERM
        DTARAY(1)=0.0
        DTARAY(2)=0.0
        CALL CBSET(CB1,CBCW,CWSTD,CBAZR,DTARAY,IER)
        CALL VSMA(CB1)


400     IF(IPRE.NE.1)GOTO 406
        CALL CBSET(CB1,CBAXR,B,CBAAMM,PREM,IER)
        CALL VLDR(CB1)                  ;LOAD EMPHASIS VECTOR
        CALL CBSET(CB1,CBAZR,DTARAY,CBAYR,DTARAY,IER)
        CALL VMRA(CB1)                  ;MULTIPLY VECTORS
C****** GET ENERGY IN SPEECHFILE
```

```
406     CALL CBSET(CB1,CBAAMM,SUME,CBAXR,DTARAY,IER)
        CALL VSER(CB1)                     ;SUM VECTOR ELEMENTS
C****** SQRT SPECTRAL COMPONENTS
        DO 410 J4=1,LM
410     DTARAY(J4)=SQRT(DTARAY(J4))
C****** NORMALIZE ENERGY IN SPECTRAL FILE TO A CONSTANT
C       OR DO SOMETHING ELSE
        IF(NORM.EQ.0)GOTO 418
        FACTOR=10000/SQRT(SUME)
        GOTO 420
418     FACTOR=10000000/SUME
420     CONTINUE
        CALL CBSET(CB1,CBSCR,FACTOR,CBAZR,B,IER)
        CALL VMRS(CB1)                     ;MULTIPLY VECTOR BY SCALER
C****** LOAD DTARAY WITH NEXT 128 PNTS
        J1=J1+LM                  ;MOVE OVER 64 PNTS
        IF(J1.EQ.1025)GOTO 429  ;NEED TO REMAP TO NEXT EXT MEM BLK
        IF(J1.EQ.961)GOTO 422   ;ONLY 64 PNTS LEFT
        J3=0
        DO 421 J2=J1,J1+127
        J3=J3+1
421     DTARAY(J3)=IDSP(J2)
        GOTO 429
422     J3=0
        DO 423 J2=J1,1024                  ;LOAD LAST 64 PNTS OF CURRENT
        J3=J3+1                            ;MEMORY BLOCK
423     DTARAY(J3)=IDSP(J2)
        IF(J6.GT.DUMMY(1))GOTO 425         ;LAST MEM BLOCK
        CALL APMAP(IDSP,J6,-1,IER)
        DO 424 J2=65,128                   ;LOAD 1ST 64 PNTS OF NEXT
        J3=J2-64                                   ;MEMORY BLOCK
424     DTARAY(J2)=IDSP(J3)
        CALL APMAP(IDSP,J6-1,-1,IER)
        GOTO 429
425     DO 427 J3=65,128                   ;NEED FUDGE LAST 64 PNTS
427     DTARAY(J3)=DTARAY(64)*EXP(64-J3)
C****** LOAD NORMALIZED SPECTRUM INTO IDOB
429     COUNT=COUNT+1
        COUNTDOWN=COUNTDOWN-1
        IDOB(JAOUT)=SQRT(SUME)
        J3=1
        DO 430 J2=JAOUT+1,JAOUT+63         ;LOADING 64 PNTS
        J3=J3+1
430     IDOB(J2)=B(J3)
        JAOUT=JAOUT+LM
        IF(JAOUT.NE.1025)GOTO 435
        JAOUT=1
        J5=J5+2
        CALL APMAP(IDOB,J5,-2,IER)
435     CONTINUE
440     FORMAT(5G12.4)
        IF(COUNTDOWN.LE.LASTCD)GOTO 1000
700     CONTINUE
750     IF(J1.LE.961)GOTO 305
```

```
800     GOTO260

1000    RETURN
        END
```

```
C****** SUBROUTINE S64  NOTE:  THIS ROUTINE IS FOR FORTRAN 5 !!
C       THIS SUBROUTINE CALLS ARRAY PROCESSOR ROUTINES TO CALCULATE
C       FFT'S.  THE WINDOW SIZE CAN BE ANY MULTIPLE OF 2 IN THE RANGE
C       [16,1024].  ALTHOUGH DESIGNED FOR SPEECH ANALYSIS, THIS
C       ROUTINE IS FLEXIBLE.
C       BY:  CAPT DAN MARTIN
C       DATE:  9/21/82
C       SUBJ.  ACOUSTIC ANALYSIS
C       THIS ROUTINE IS CALLED BY DRSQ WHICH SETS UP THE NECESSARY FILE
C       INPUT/OUTPUT OPTIONS.  THE INPUT TO THIS ROUTINE RESIDES IN
C       EXTENDED MEMORY AND THE OUTPUT IS DEPOSITED THERE.
C       S64 DOES THE FOLLOWING:
C               1)  INITIALIZES PARAMETERS
C               2)  MAPS OUTPUT ARRAY IDOB
C               3)  MAPS INPUT ARRAY IDSP
C               4)  LOADS TIME SLICE INTO DTARAY
C               5)  WINDOWS TIME SLICE
C               6)  CALCULATES FFT
C               7)  ZEROS-OUT 1ST AND LAST ELEMENTS
C               8)  EMPHASIZES SPECTRUM
C               9)  NORMALIZES SPECTRUM
C               10)  LOADS SPECTRUM INTO IDOB
C               11)  REMAPS IDOB IF NECESSARY
C               12)  REMAPS IDSP IF NECESSARY
C       FOR MORE INFORMATION, SEE MY THESIS OR THE USERS MANUAL.


        SUBROUTINE S64
        INCLUDE "ARRAYP:F5APS.FR"     ; APS PARAMETER FILE

C******     Set up variables to be used.
        PARAMETER LM=FLEN
        PARAMETER HLM=LM/2
        PARAMETER KLM=HLM-1
        PARAMETER NDP=1024
        PARAMETER MAXLM=512
        REAL DTARAY,B,PREM,WIND,SFREQ
        REAL SUME,FACTOR,IDOB
        INTEGER CB1,FLEN,FILE1,FILE2,FLAG,CNTBR,IDSP,INITMEM,SXMEM
        INTEGER IHAM,IPRE,SKIP,COUNT,SMOOTH,WRTD,NBLKS,XMEM,DUMMY
        INTEGER CSPCT,LTSTDO,COUNTDOWN,NORM
        COMMON /APM/ DTARAY(NDP),B(NDP)          ;AP MEMORY
        COMMON / VALS / IDSP(NDP),IDOB(NDP),IHAM,IPRE,WIND(MAXLM)
        COMMON / VALS / PREM(MAXLM),INITMEM,SXMEM,DUMMY(304),COUNTDOWN
        COMMON / VALS / LTSTDO
        COMMON / VALT / SKIP,COUNT,SMOOTH,WRTD,NBLKS,XMEM,SFREQ,FLEN
        COMMON / VALT / FILE1(13),FILE2(13),FLAG,CNTBR,ISXMEM
        COMMON / VALT / JAOUT,LASTCD
        COMMON / VALV / CB1(0:CBMAX)
        COMMON / VALU / NORM

C****** INITIALIZE APS AND APM MAP
        IF(SKIP.EQ.0)GOTO 250
        CALL APINIT(NIL,DTARAY,9,INITMEM,IER)
        CALL APMAP(DTARAY,0,4,IER)
250     J5=9                    ;MEM BLOCK
260     CONTINUE
        CALL APMAP(IDSP,J5,-1,IER)               ;INPUT ARRAY
        CALL APMAP(IDOB,J5,-1,IER)               ;OUTPUT ARRAY

        JAOUT=1                          ;POINTER IN OUTPUT VECTOR
C****** ITERATE DOING FFT'S BY TIME-SLICE
        DO 700 J1=1,NDP,LM
C       INPUT SEQUENCE IS INTEGER, OUTPUT IS REAL.  SO OUTPUT
C       TAKES PLACE OF INPUT ELEMENT FOR ELEMENT
        DO 300 J2=J1,J1+LM-1
        J3=J2-(J1-1)
300     DTARAY(J3)=IDSP(J2)


C****** WINDOW VECTOR DTARAY
        IF(IHAM.NE.1)GOTO 310


        CALL APSETL(LM,IER)
        CALL CBSET(CB1,CBAXR,B,CBAAMM,WIND,IER)
        CALL VLDR(CB1)                   ;LOAD WINDOW VECTOR
        CALL CBSET(CB1,CBAXR,DTARAY,CBAYR,B,CBAZR,DTARAY,IER)
        CALL VMRA(CB1)                   ;MULTIPLY VECTORS

  310   CONTINUE
C****** CALCULATE FFT OF VECTOR DTARAY

        CALL APSETL(HLM,IER)

        CALL CBSET(CB1,CBAXC,DTARAY,CBCW,CWDFT,IER)
        CALL VFFTC(CB1)
        CALL VBRC(CB1)
        CALL VFFTR(CB1)
C****** DTARAY CONTAINS 1ST HLM+1 SPECTRUM COEFFICIENTS
C       NEED ZERO-OUT 1ST ELEMENT, I.E., ZERO-HERTZ TERM
C       ALSO NEED ZERO-OUT 33RD ELEMENT, I.E., 4KHZ TERM
        DTARAY(1)=0.0
        DTARAY(2)=0.0
        CALL CBSET(CB1,CBCW,CWSTD,CBAZR,DTARAY,IER)
        CALL VSMA(CB1)          ;GET SQUARE MAGNITUDE

400     IF(IPRE.NE.1)GOTO 406
        CALL CBSET(CB1,CBAXR,B,CBAAMM,PREM,IER)
        CALL VLDR(CB1)          ;LOAD ARRAY
        CALL CBSET(CB1,CBAZR,DTARAY,CBAYR,DTARAY,IER)
        CALL VMRA(CB1)                   ;MULTIPLY VECTORS


406     CALL CBSET(CB1,CBAAMM,SUME,CBAXR,DTARAY,IER)
        CALL VSER(CB1)                   ;SUM ELEMENTS
C****** SORT SPECTRAL COMPONENTS
        DO 420 J4=2,HLM
```

```
420     DTARAY(J4)=SQRT(DTARAY(J4))
C****** NORMALIZE ENERGY IN SPECTRAL FILE TO A CONSTANT
C       OR DO SOMETHING ELSE
        IF(NORM.EQ.0)GOTO 422
        FACTOR=10000/SQRT(SUME)
        GOTO 425
422     FACTOR=10000000/SUME
425     CONTINUE
        CALL CBSET(CB1,CBSCR,FACTOR,CBAZR,DTARAY,IER)
        CALL VMRS(CB1)                     ;MULTIPLY BY SCALER
C****** LOAD NORMALIZED SPECTRUM INTO IDOB
        COUNTDOWN=COUNTDOWN-1
        COUNT=COUNT+1
        IDOB(JAOUT)=SQRT(SUME)  ;LOAD ENERGY
        J3=1
        DO 430 J2=JAOUT+1,JAOUT+KLM      ;LOADING ONLY HLM POINTS
        J3=J3+1
430     IDOB(J2)=DTARAY(J3)
440     FORMAT(5G12.4)

        IF(COUNTDOWN.LE.LASTCD)GOTO 1000
700     JAOUT=JAOUT+HLM
800     J5=J5+1
        GOTO260

1000    RETURN
        END
```

FILE:	DSTA and DSTN
LANGUAGE:	FORTRAN 5
DATE:	September 21, 1982
AUTHOR:	D. Martin
SUBJECT:	Acoustic Analysis
CALLING SEQUENCE:	DSTA or DSTN
DATE OF LAST REVISION:	September 21, 1982

PURPOSE:

These programs compute distances between observations and phonets. DSTA fills the output disk file with all the distances between the specified observations and phonets. DSTN fills the output disk file with one unit for each specified observation. Each unit contains the observation number, observation energy, worst phonet match, maximum distance, and a selected number of ordered pairs: (phonet number, next best distance).

DESCRIPTION:

Location: DP4:BRATCHET

Size:	DSTN.FR	12912 bytes
	DSTN.RB	17402 bytes
	DSTA.FR	12699 bytes
	DSTA.RB	17488 bytes

PROGRAM USE:

These programs are called by DRVR and call CHOOS, GFDB, CHKO, and CHKJ. They use the Array Processor to compute distances between observations and phonets from files calculated by DRSQ. The distinction between observation le and phonet file is arbitrary in format; any file can run against itself. The distinction in the way these ıtines handle files is that distances to all phonets e computed for each observation.

The main difference between these routines is that ile DSTA outputs every computed distance, DSTN selects e best matches and outputs the observation number, energy, d worst match as well. Tables 4 and 5 contain the output le formats.

Two distance rules are selectable: M1 and M2. M1 is ıkowski One distance:

$$D_{ij} = K \sum_{l=1}^{N} \left| X_{il} - Y_{jl} \right|$$

ere:

D_{ij} = distance between the i^{th} observation and the j^{th} phonet.

K = constant scale factor.

X_{il} = l^{th} component of the i^{th} observation.

Y_{jl} = l^{th} component of the j^{th} phonet.

is the Minkowski Two distance:

$$D_{ij} = K \sqrt{\sum_{l=1}^{N} (X_{il} - Y_{jl})^2}$$

ere:

D_{ij} = distance between the i^{th} observation and the j^{th} phonet.

K = constant scale factor.

X_{il} = l^{th} component of the i^{th} observation.

ELEMENT NUMBER	ASSIGNMENT
1	D(i, j)
2	D(i, j+1)
3	D(i, j+2)
.	.
.	.
.	.
K-j+1	D(i, K)
K-j+2	D(i+1, j)
K-j+3	D(i+1, j+1)
.	.
.	.
.	.
2K-j+1	D(i+1, K)
.	.
.	.
.	.
(l+1)K-j+1	D(i+l, K)

where: i = starting observation number specified.

j = starting phonet number specified.

i+l = last observation number specified.

K = last phonet specified.

TABLE 4. DSTA Output File

ELEMENT NUMBER	ASSIGNMENT
1	Number of first observation specified.
2	Energy of first observation specified.
3	Number of worst phonet choice.
4	Largest distance.
5	Number of best phonet choice.
6	Minimum distance.
7	Number of next best phonet choice.
8	Next minimum distance.
.	.
.	.
.	.
4+2N	N^{th} minimum distance.
4+2N+1	Number of next observation specified.
.	.
.	.
.	.
K(4+2N)	N^{th} minimum distance to K^{th} observation.

TABLE 5. DSTN Output File

Y_{jl} = lth component of the jth phonet

The observation energy is not included in either distance computation. Also, the distance is checked for overflow before being truncated to an integer and written to disk. If the distance is greater than 32767, it is hard limited to that value. The scale factor, K in the computations, is adjustable to obtain a satisfactory range of distance measures. The flowchart in Figure 74 applies to both DSTA and DSTN. Distance file header assignments are in Table 6.

LIST OF VARIABLES:

VARIABLE	TYPE	PURPOSE
XMEM	Integer	Number of extended memory blocks to use.
OUTD	Integer	Output option switch.
FILE2	Integer Array	Holds observation file name.
FILE1	Integer Array	Holds phonet file name.
FILE3	Integer Array	Holds output distance file name.
STS1	Integer	First observation to do.
STSL	Integer	Last observation to do.
PTS1	Integer	First phonet to do.
PTSL	Integer	Last phonet to do.
HEADER	Integer Array	Holds header.
NSTSTB	Integer	Number of observations to do.
NPSTS	Integer	Number of words in observation.

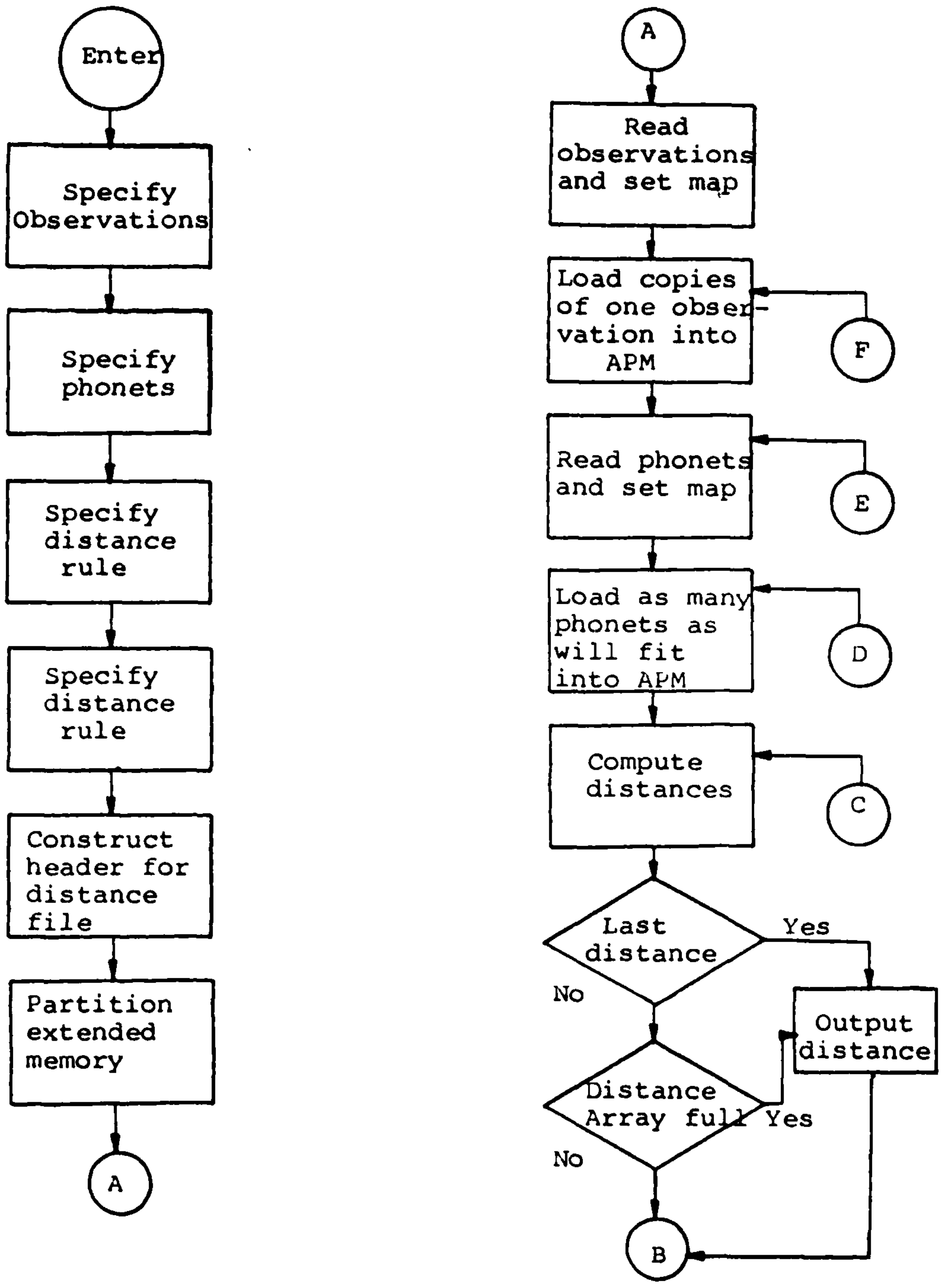


FIGURE 74. Flowchart of DSTA and DSTN

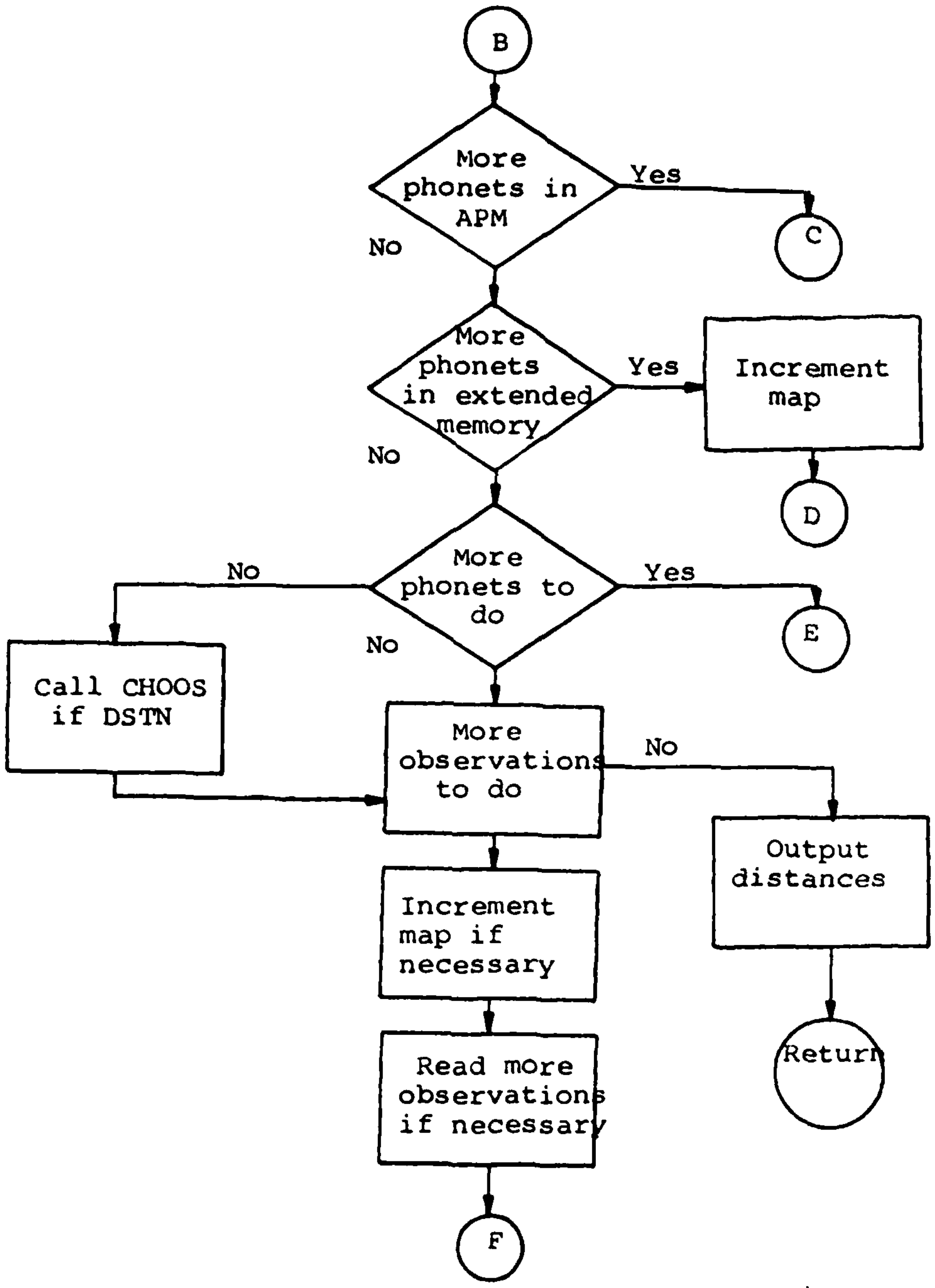


FIGURE 74 CONTINUTED. Flowchart of DSTA and DSTN

ELEMENT NUMBER	CONTENT
1-13	Distance file name.
14-26	Observation file name.
27-39	Phonet file name.
40	Number of first observation time slice to do.
41	Number of last observation time slice to do.
42	Number of first phonet time slice to do.
43	Number of last phonet time slice to do.
44	Number of disk block that holds first observation to do.
45	Number of disk block that holds first phonet to do.
46	Switch: 4 = observation and phonet files identical/0 = different.
47	NCHOICES.
48	Number of observation time slices to do.
49	Number of phonet time slices to do.
50	Number of elements per time slice.
51	Number of extended memory block used less those in window.
52-256	Unused.

TABLE 6. Distance File Header

VARIABLE	TYPE	PURPOSE
IBO	Integer	Holds the first observation number.
FSPFSQ	Integer	Switch which selects phonet file.
NPTSTB	Integer	Number of phonets to do.
NLDTB	Integer	Number of distances to compute.
JCTLD	Integer	Counts distances computed.
IBP	Integer	Holds the first phonet number.
FDSTR	Integer	Switch which selects distance rule.
NPFS	Integer	Number of required memory blocks for observations.
NPFP	Integer	Number of required memory blocks for phonets.
NBFPH	Integer	Number of extended memory blocks to use for phonets.
NBFOBS	Integer	Number of extended memory blocks to use for observations.
NBFDIST	Integer	Number of extended memory blocks to use for distances.
OBSMAPCNT	Integer	First extended memory block used for observations.
FONMAPCNT	Integer	First extended memory block used for phonets.
DISTMAPCNT	Integer	Extended memory block used for distances.
FONBANGCNT	Integer	Counts phonets.
OBSLEFT	Integer	Specifies observation points left.

VARIABLE	TYPE	PURPOSE
FONLEFT	Integer	Specifies phonet points left.
NPPTS	Integer	Number of words in phonet.
WKOBS	Real Array	Array Processor memory array.
WKFON	Real Array	Array Processor memory array.
DIST	Integer Array	Output distance file.
JDISTNBR	Integer	Counts phonets in WKFON.
JOUTDIST	Integer	Counts distances in array DIST.
NDBFDIST	Integer	Holds disk block to write distances to.
NQBFOBS	Integer	Number of quarter blocks for observations.
NQBFPH	Integer	Number of quarter blocks for phonets.
CNTOBS	Integer	First disk block of observations to get.
IFBC	Integer	First data point in disk block.
ICNTOBS	Integer	Counts observation disk blocks read.
CNTFON	Integer	First disk block of phonets to get.
IFBD	Integer	First data point in disk block.
ICNTFON	Integer	Counts phonet disk blocks read.
NFONLD	Integer	Number of phonets in next map.
NPLTB	Integer	Number of phonets left to do.

VARIABLE	TYPE	PURPOSE
FONLEFT	Integer	Number of points in WKFON to fill.
JOUTBLKS	Integer	Number of distance quarter blocks to write to disk.
JOUTDIST	Integer	Counts distances.
FON	Real Array	Mapped through extended memory to get phonets.
OBS	Real Array	Mapped through extended memory to get observations.

SWITCH SETTINGS:

SWITCH		SETTINGS
OUTD	=	0 for disk output only
	=	10 for screen output as well
	=	12 for printed output as well
FSPFSQ	=	4 to run the observation file against itself
	=	3 to use separate phonet file
FDSTR	=	1 for M1 distance rule
	=	2 for M2 distance rule

RELATED PROGRAMS:

PLTA, PLTO, PLTS, PLTT, PLTN, DRVR, S64, S128, CHOOS, FCTR, GFDB, CHKO, CHKJ, GRPH2.

```
C****** ROUTINE DSTN  NOTE:  THIS ROUTINE IS FOR FORTRAN  5 ! !
C       THIS SUBROUTINE CALLS ARRAY PROCESSOR ROUTINES AND USES EXTENDED
C       MEMORY (14 BLOCKS OR MORE) TO COMPUTE A SPECIFIED NUMBER OF BEST
C       DISTANCES, BEST BEING SMALLEST, BETWEEN SPECIFIED OBSERVATIONS
C       AND SPECIFIED PHONETS.
C       BY:  CAPT DAN MARTIN
C       DATE: 9/21/82
C       SUBJ: ACOUSTIC ANALYSIS
C       THIS ROUTINE IS CALLED BY DRVR AND CALLS ROUTINES CHOOS, CHKO, CHKJ,
C       AND GFDB.  ONE OF TWO DISTANCE RULES, M1 OR M2, CAN BE SELECTED.
C       INPUT FILES ARE SPECTRAL FILES OUTPUT BY DRSQ.  THE OUTPUT FILE IS
C       AN INTEGER FILE OF CONSECUTIVE UNITS, EACH UNIT BEING 4+2N POINTS
C       LONG, N BEING THE NUMBER OF BEST PHONET CHOICES SPECIFIED.  IN EACH
C       UNIT, THE FIRST ELEMENT IS THE OBSERVATION NUMBER, THE SECOND
C       ELEMENT IS THE OBSERVATION ENERGY, THE THIRD AND FOURTH ARE AN
C       ORDERED PAIR: (PHONET NUMBER,MAX DISTANCE), AND THE REMAINING
C       POINTS ARE N-ORDERED PAIRS, ORDERED BEST-TO-WORST: (PHONET NUMBER,
C       DISTANCE).
C       THIS ROUTINE ACCOMPLISHES THE FOLLOWING TASKS:
C               1)  GET OBSERVATION FILE
C               2)  SPECIFY OBSERVATIONS-FIRST AND LAST-TO DO
C               3)  GET PHONET FILE
C               4)  SPECIFY PHONETS-FIRST AND LAST-TO DO
C               5)  GET DISTANCE RULE
C               6)  GET FILE NAME FOR OUTPUT DISTANCES
C               7)  CONSTRUCT OUTPUT HEADER
C               8)  PARTITION EXTENDED MEMORY BETWEEN DISTANCES,
C               OBSERVATIONS, AND PHONETS
C               9)  READ OBSERVATIONS INTO MEMORY
C               10) GET ONE OBSERVATION
C               11)  READ PHONETS INTO MEMORY
C               12)  GET PHONETS TO DO
C               13)  GET ALL DISTANCES FOR CURRENT PHONET
C               14)  CALL CHOOS TO FILL OUTPUT FILE
C               15)  GET NEXT OBSERVATION
C               16)  GO TO STEP 11 UNTIL ALL OBSERVATIONS ARE DONE
C               17)  OUTPUT DISTANCES ACCUMULATED
C       FOR MORE INFORMATION, SEE THE USERS MANUAL OR MY THESIS.

        SUBROUTINE DSTN
        OVERLAY ODSTN
        INCLUDE"ARRAYP:F5APS.FR"
        PARAMETER NDP=1024
        PARAMETER NDPM1=1023
        PARAMETER QTRBLK=256
        REAL WKFON,WKOBS,NPFSR,NPFPR,FACTOR,FON,OBS,HOLDIST
        REAL MAXDIST,MINDIST
        INTEGER FSPFN,FILE2,HEADER,STS1,STSL,NSTSTB,NPSTS,NPFS
        INTEGER DIST,FSPFSQ,FILE1,PTS1,PTSL,JOUTDIST
        INTEGER NPTSTB,NPPTS,NPFP,FDSTR,FILE3,OUTD,FLAG,CB1,SKIP
        INTEGER XMEM,NBFDIST,NBFOBS,NBFPH,CNTOBS,CNTFON,OBSMAPCNT
        INTEGER FONMAPCNT,FONBANCCNT,OBSLEFT,NQBFOBS,NQBFPH,NQBFDIST
        INTEGER DISTMAPCNT,OBSBANCCNT,FONLEFT,ICNTOBS,IFBC
        INTEGER NPLTB,JFONMAPCNT,JOBSMAPCNT,DISTMAPCNTMAX,NDBFDIST
        INTEGER JDISTNBR,JCNTOBS,JCNTFON,ICNTFON,NFONLD,INITMEM

        COMMON / APM / WKOBS(NDP),WKFON(NDP)
        COMMON / VALS / OBS(NDP),FON(NDP),DIST(NDP),OBSLEFT,FSPFN
        COMMON / VALS / INITMEM,FILE3(13),HEADER(QTRBLK),NPFSR,NPFPR
        COMMON / VALS / PTS1,PTSL,NPTSTB,NPPTS,NPFP,FDSTR,OUTD,NBFDIST
        COMMON / VALS / NBFOBS,NBFPH,CNTOBS,CNTFON,OBSMAPCNT,FONMAPCNT
        COMMON / VALS / FONBANGCNT,NQBFOBS,NQBFPH,NQBFDIST,DISTMAPCNT
        COMMON / VALS / ICNTOBS,ICNTFON,IFBC,JFONMAPCNT,NFONLD
        COMMON / VALS / OBSBANGCNT,FONLEFT,NPLTB,JOBSMAPCNT,NDBFDIST
        COMMON / VALS / DISTMAPCNTMAX,JOUTDIST,JDISTNBR,JCNTOBS,JCNTFON
        COMMON / VALT / SKIP,STSL,NSTSTB,NPSTS,NPFS,XMEM,FACTOR,FSPFSQ
        COMMON / VALT / FILE1(13),FILE2(13),FLAG,STS1,ISXMEM
        COMMON / VALV / CB1(0:CBMAX)
        COMMON / VALU / HOLDIST(NDP),MAXDIST,MINDIST,MAXDISTLOC,MINDISTLOC
        COMMON / VALU / NCHOICES,NBRPHONES

        TYPE"****YOU ARE NOW IN ROUTINE DSTN.FR****"

50      TYPE"*YOU HAVE",ISXMEM,"  1K BLOCKS OF EXT MEMORY ACCESSABLE"
        XMEM=ISXMEM
3       TYPE"*ENTER OUTPUT DEVICE *"
        ACCEPT"(0=NONE/10=TERMINAL/12=PRINTER):  ",OUTD
        IF(OUTD.EQ.0.OR.OUTD.EQ.10.OR.OUTD.EQ.12)GOTO 5
        TYPE"ERROR: INVALID VALUE FOR OUTD:  ",OUTD
        GOTO 3

C****** GET OBSERVATION FILE NAME
5       ACCEPT"**ENTER NAME OF FILE THAT HOLDS OBSERVATION TIME-SLICES: "
        READ(11,10)FILE2(1)
10      FORMAT(S13)
12      CALL OPEN(4,FILE2,2,IER)
        IF(IER.NE.1)TYPE"ERROR ON OPEN OF SPECTRAL FILE, IER= ",IER
        GOTO 18
15      CONTINUE
C****** GET NUMBERS OF 1ST & LAST TIME-SLICES OF SPECTRUM
18      CALL RDBLK(4,0,HEADER,1,IER)
        IF(OUTD.EQ.0)GOTO 19
        WRITE(OUTD,9001)(HEADER(I5),I5=1,13)
19      TYPE"1ST SPECTRAL TIME-SLICE * IS:",HEADER(55)
        TYPE"LAST SPECTRAL TIME-SLICE * IS:",HEADER(56)
20      ACCEPT"***ENTER 1ST SPECTRAL TIME-SLICE * TO BANG:  ",STS1
        IF(STS1.GE.HEADER(55))GOTO 22
        TYPE"ERROR: 1ST TIME-SLICE TOO SMALL!  MUST BE GREATER THAN OR"
        TYPE"EQUAL TO",HEADER(55)
        GOTO 20
22      ACCEPT"***ENTER LAST SPECTRAL TIME-SLICE * TO BANG:  ",STSL
        IF(STSL.LE.HEADER(56))GOTO 23
        TYPE"ERROR: LAST TIME-SLICE TOO LARGE!  MUST BE LESS THAN OR"
        TYPE"EQUAL TO",HEADER(56)
        GOTO 22
23      IF(STS1.LT.STSL)GOTO 24
        TYPE"ERROR: 1ST TIME-SLICE LARGER THAN LAST!"
```

```
      GOTO 20
C****** GET # OF TS'S TO BANG AND AMOUNT OF MEMORY REQUIRED
24    NSTSTB=STSL-STS1+1        ;# TS'S TO BANG
      NPSTS=HEADER(57)*2        ;# WORDS PER TS
      IBO=HEADER(55)
C****** GET PHONET FILE
      TYPE"*DO YOU WANT TO BANG THE SPECTRAL FILE AGAINST ITSELF?"
      ACCEPT"ENTER (4=YES/3=NO):  ",FSPFSQ
      IF(FSPFSQ.EQ.4)GOTO 37
35    ACCEPT"***ENTER PHONET FILENAME:  "
      READ(11,10)FILE1(1)
36    CALL OPEN(3,FILE1,2,IER)
      CALL CHECK(IER)
      CALL RDBLK(3,0,HEADER,1,IER)
      CALL CHECK(IER)
      IF(OUTD.EQ.0)GOTO 39
      WRITE(OUTD,9001)(HEADER(I5),I5=1,13)
      GOTO 39
C****** GET THE 1ST & LAST PHONETS
37    DO 38 J1=1,13
38    FILE1(J1)=FILE2(J1)
39    TYPE"1ST PHONET TS# IS:",HEADER(55)
      TYPE"LAST PHONET TS# IS: ",HEADER(56)
41    ACCEPT"***ENTER 1ST PHONET TS TO DO:  ",PTS1
      IF(PTS1.GE.HEADER(55))GOTO 42
      TYPE"ERROR: 1ST TIME-SLICE IS TOO SMALL!  MUST BE GREATER"
      TYPE"THAN OR EQUAL TO",HEADER(55)
      GOTO 41
42    ACCEPT"***ENTER LAST PHONET TS TO DO:  ",PTSL
      IF(PTSL.LE.HEADER(56))GOTO 43
      TYPE"ERROR: LAST TIME-SLICE TOO BIG!  MUST BE SMALLER THAN"
      TYPE"OR EQUAL TO",HEADER(56)
      GOTO 42
43    IF(PTS1.LE.PTSL)GOTO 44
      TYPE"ERROR: 1ST TIME-SLICE LARGER THAN LAST!"
      GOTO 41
44    ACCEPT" ENTER NUMBER OF DISTANCE CHOICES TO GET: ",NCHOICES
C****** GET NUMBER OF PHONET TS'S TO BANG AND AMOUNT OF MEMORY
C       THAT WILL TAKE
      NPTSTB=PTSL-PTS1+1        ;# TS'S TO BANG
      NLDTD=NSTSTB*(4+2*NCHOICES)           ;# DISTANCES TO GET
      IBP=HEADER(55)
C****** VERIFY # PNTS IN PHONET TIME-SLICES MATCH # PNTS IN SPECTRAL TS'S
      NPPTS=HEADER(57)*2
      IF(NPPTS.EQ.NPSTS)GOTO 60
      TYPE"ERROR: # PNTS IN PHONETIC TIME-SLICE MUST MATCH # PNTS"
      TYPE"IN SPECTRAL TIME-SLICE.  AS SPECIFIED,"
      TYPE"# PNTS IN PTS=",NPPTS
      TYPE"# PNTS IN STS=",NPSTS
      GOTO 35
60    CONTINUE
C****** GET DISTANCE RULE
      ACCEPT"*SELECT DISTANCE RULE (1=M1/2=M2):  ",FDSTR
C****** GET DISTANCE FILE
```

```
C****** GET NAME OF FILE TO STASH DISTANCES INTO
65      ACCEPT"***ENTER NAME OF FILE TO RECIEVE DISTANCES: "
        READ(11,10)FILE3(1)
70      CALL OPEN(2,FILE3,2,IER)
        IF(IER.NE.1)TYPE"ERROR ON OPEN OF DISTANCE FILE, IER= ",IER
        IF(IER.NE.13)GOTO 75
        TYPE"FILE DOES NOT EXIST.  WILL CREATE IT FOR YOU."
        CALL CFILW(FILE3,2,IER1)
        IF(IER1.NE.1)TYPE"ERROR ON FILE CREATION. IER1= ",IER1
        GOTO 70
75      CONTINUE
C****** STORE FILENAMES IN HEADER OF DISTANCE FILE
        DO 100 J1=1,QTRBLK
100     HEADER(J1)=0.0
        DO 110 J1=1,13
        HEADER(J1)=FILE3(J1)
        J2=J1+13
        HEADER (J2)=FILE2(J1)
        J3=J1+26
110     HEADER(J3)=FILE1(J1)
        HEADER(40)=STS1
        HEADER(41)=STSL
        HEADER(42)=PTS1
        HEADER(43)=PTSL
        HEADER(46)=FSPFSQ       ;SWITCH INDICATES IF OBS AND PHONS ARE SAME
        HEADER(47)=NCHOICES     ;# DISTANCE CHOICES
        HEADER(48)=NSTSTB       ;# OBS TIME SLICES
        HEADER(49)=NPTSTB       ;# PHON TIME SLICES
        HEADER(50)=NPPTS        ;# ELEMENTS PER TIME-SLICE
C****** DIVVY-UP EXTENDED MEMORY BETWEEN OBSERVATION SPECTRUM,
C       PHONET SPECTRUM, AND THE DISTANCE FILES.
111     NPFS=2+2*IFIX(FLOAT(NSTSTB)*NPSTS/(2*NDP)) ; MEM BLKS FOR OBS FILE
        NPFP=2+2*IFIX(FLOAT(NPTSTB)*NPPTS/(2*NDP)) ; MEM BLKS FOR PHON FILE
114     CONTINUE
116     J1=IFIX((XMEM-3)/2)*2           ;NEED EVEN # MEM BLKS
        IF(NPFP.LT.J1)NBFPH=NPFP
        IF(NPFP.GE.J1)NBFPH=J1
        J1=IFIX((XMEM-1-NBFPH)/2)*2
        IF(NPFS.LT.J1)NBFOBS=NPFS
        IF(NPFS.GE.J1)NBFOBS=J1
        NBFDIST=1
        IF(NBFOBS.GT.0)GOTO 118
        TYPE"ERROR: INSUFFICIENT EXTENDED MEMORY AVAILABLE."
        TYPE"NEED AT LEAST 14 1K BLOCKS."
        CALL VMEM(XMEM,IER)
        TYPE"YOU HAVE",XMEM,"  1K BLOCKS OF EXTENDED MEMORY AVAILABLE."
        ACCEPT"*HOW MANY DO YOU WANT?  ",XMEM
        XMEM=XMEM-9
        GOTO 114
118     CONTINUE
        HEADER(51)=XMEM
C****** INITIALIZE TO 1ST AVAILABLE EXT MEM BLK
        OBSMAPCNT=9             ;1ST EXT MEM BLK USED FOR OBSERVATIONS
        FONMAPCNT=OBSMAPCNT+NBFOBS              ;1ST EXT MEM BLK
```

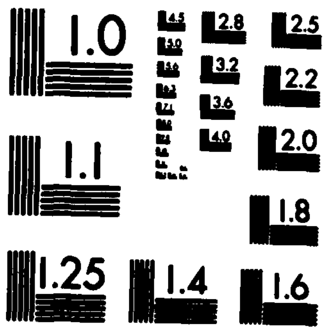
```
;****** ROUTINE DSTA  NOTE:  THIS ROUTINE IS FOR FORTRAN 5 ! !
;       THIS SUBROUTINE CALLS ARRAY PROCESSOR ROUTINES AND USES EXTENDED
;       MEMORY (14 MEMORY BLOCKS OR MORE) TO COMPUTE ALL DISTANCES
;       BETWEEN EACH OBSERVATION SPECIFIED AND EACH PHONET SPECIFIED.
;       BY:  CAPT DAN MARTIN
;       DATE:  9/21/82
;       SUBJ:  ACOUSTIC ANALYSIS
;       THIS ROUTINE IS CALLED BY DRVR AND CALLS CHKO AND GFDB.  INPUT
;       FILES ARE SPECTRAL FILES OUTPUT BY DRSQ.  THE OUTPUT FILE IS AN
;       INTEGER FILE OF CONSECUTIVE UNITS.  EACH UNIT CONTAINS DISTANCES
;       BETWEEN AN OBSERVATION AND EACH PHONET, THE K-TH ELEMENT OF THE
;       J-TH UNIT IS THE DISTANCE BETWEEN THE J-TH OBSERVATION AND THE
;       K-TH PHONET.
;       THIS ROUTINE ACCOMPLISHES THE FOLLOWING TASKS:
;               1)  GET OBSERVATION FILE
;               2)  SPECIFY OBSERVATIONS-FIRST AND LAST-TO DO
;               3)  GET PHONET FILE
;               4)  SPECIFY PHONETS-FIRST AND LAST-TO DO
;               5)  GET DISTANCE RULE
;               6)  GET FILE NAME FOR OUTPUT DISTANCES
;               7)  CONSTRUCT OUTPUT HEADER
;               8)  PARTITION EXTENDED MEMORY BETWEEN DISTANCES,
;               OBSERVATIONS, AND PHONETS
;               9)  READ OBSERVATIONS INTO MEMORY
;               10) GET ONE OBSERVATION
;               11)  READ PHONETS INTO MEMORY
;               12)  GET PHONETS TO DO
;               13)  GET A DISTANCE
;               14)  OUTPUT DISTANCES WHEN 1024 HAVE BEEN ACCUMULATED
;               15)  GET MORE PHONETS UNTIL ALL ARE DONE
;               16)  GET NEXT OBSERVATION
;               17)  GO TO STEP 11 UNTIL ALL OBSERVATIONS ARE DONE
;               18)  OUTPUT DISTANCES ACCUMULATED
;       FOR MORE INFORMATION, SEE THE USERS MANUAL OR MY THESIS.

        SUBROUTINE DSTA
        OVERLAY ODSTA
        INCLUDE"ARRAYP:F5APS.FR"
        PARAMETER NDP=1024
        PARAMETER NDPM1=1023
        PARAMETER WTRBLK=256
        REAL WKFON,WKOBS,NLDTD,JCTLD,NPFSR,NPFPR,FACTOR,FON,OBS
        INTEGER FSPFN,FILE2,HEADER,STS1,STSL,NSTSTB,NPSTS,NPFS
        INTEGER DIST,FSPFSO,FILE1 PTS1,PTSL,JOUTDIST
        INTEGER NPTSTB,NPPTS,NPFP,FDSTR,FILE3,OUTD,FLAG,CB1,SKIP
        INTEGER XMEM,NBFDIST,NBFOBS,NBFPH,CNTOBS,CNTFON,OBSMAPCNT
        INTEGER FONMAPCNT,FONBANGCNT,OBSLEFT,NOBFOBS,NOBFPH,NOBFDIST
        INTEGER DISTMAPCNT,OBSBANGCNT,FONLEFT,ICNTOBS,IFBC,GSPCT
        INTEGER NPLTB,JFONMAPCNT,JOBSMAPCNT,DISTMAPCNTMAX,NDBFDIST
        INTEGER JDISTNBR,JCNTOBS,JCNTFON,ICNTFON,NFONLD,INITMEM


        COMMON / APM / WKOBS(NDP),WKFON(NDP)
        COMMON / VALS / OBS(NDP),FON(NDP),DIST(NDP),OBSLEFT,FSPFN
```

AD-A125 360 COMPUTER RECOGNITION OF PHONETS IN SPEECH(U) AIR FORCE INST OF TECH WRIGHT-PATTERSON AFB OH SCHOOL OF ENGINEERING D L MARTIN DEC 82 AFIT/GE/EE/82D-46 3/3

UNCLASSIFIED F/G 9/2 NL

END
DATE
FILMED
4 83
DTIC

M-2



```
        COMMON / VALS / INITMEM,FILE3(13),HEADER(QTRBLK),NPFSR,NPFPR
        COMMON / VALS / PTS1,PTSL,NPTSTB,NPPTS,NPFP,FDSTR,OUTD,NBFDIST
        COMMON / VALS / NBFOBS,NBFPH,CNTOBS,CNTFON,OBSMAPCNT,FONMAPCNT
        COMMON / VALS / FONBANGCNT,NQBFOBS,NQBFPH,NQBFDIST,DISTMAPCNT
        COMMON / VALS / ICNTOBS,ICNTFON,IFBC,JFONMAPCNT,NFONLD
        COMMON / VALS / OBSBANGCNT,FONLEFT,NPLTB,JOBSMAPCNT,NDBFDIST
        COMMON / VALS / DISTMAPCNTMAX,JOUTDIST,JDISTNBR,JCNTOBS,JCNTFON
        COMMON / VALT / SKIP,STSL,NSTSTB,NPSTS,NPFS,XMEM,FACTOR,FSPFSQ
        COMMON / VALT / FILE1(13),FILE2(13),FLAG,STS1,ISXMEM
        COMMON / VALV / CB1(0:CBMAX)

        TYPE"****YOU ARE NOW IN ROUTINE DSTA.FR****"


50      TYPE"*YOU HAVE",ISXMEM,"  1K BLOCKS OF EXT MEMORY ACCESSABLE"
        XMEM=ISXMEM

3       TYPE"*ENTER OUTPUT DEVICE #"
        ACCEPT"(0=NONE/10=TERMINAL/12=PRINTER):  ",OUTD
        IF(OUTD.EQ.0.OR.OUTD.EQ.10.OR.OUTD.EQ.12)GOTO 5
        TYPE"ERROR: INVALID VALUE FOR OUTD:  ",OUTD
        GOTO 3

C****** GET OBSERVATION FILE NAME
5       ACCEPT"**ENTER NAME OF FILE THAT HOLDS OBSERVATION TIME-SLICES: "
        READ(11,10)FILE2(1)
10      FORMAT(S13)
12      CALL OPEN(4,FILE2,2,IER)
        IF(IER.NE.1)TYPE"ERROR ON OPEN OF SPECTRAL FILE, IER= ",IER
        CALL CHKO(IER)          ;IS UNIT # IN USE?
        GOTO 18
15      CONTINUE
C****** GET NUMBERS OF 1ST & LAST TIME-SLICES OF SPECTRUM
18      CALL RDBLK(4,0,HEADER,1,IER)
        CALL CHECK(IER)
        IF(OUTD.NE.0)WRITE(OUTD,9001)(HEADER(I5),I5=1,13)
19      TYPE"1ST SPECTRAL TIME-SLICE # IS:",HEADER(55)
        TYPE"LAST SPECTRAL TIME-SLICE # IS:",HEADER(56)
20      ACCEPT"***ENTER 1ST SPECTRAL TIME-SLICE # TO BANG:  ",STS1
        IF(STS1.GE.HEADER(55))GOTO 22
        TYPE"ERROR: 1ST TIME-SLICE TOO SMALL!  MUST BE GREATER THAN OR"
        TYPE"EQUAL TO",HEADER(55)
        GOTO 20
22      ACCEPT"***ENTER LAST SPECTRAL TIME-SLICE # TO BANG:  ",STSL
        IF(STSL.LE.HEADER(56))GOTO 23
        TYPE"ERROR: LAST TIME-SLICE TOO LARGE!  MUST BE LESS THAN OR"
        TYPE"EQUAL TO",HEADER(56)
        GOTO 22
23      IF(STS1.LE.STSL)GOTO 24
        TYPE"ERROR: 1ST TIME-SLICE LARGER THAN LAST!"
        GOTO 20
C****** GET # OF TS'S TO BANG AND AMOUNT OF MEMORY REQUIRED
24      NSTSTB=STSL-STS1+1      ;# TS'S TO BANG
        NPSTS=2*HEADER(57)      ;# WORDS PER TS
        IBO=HEADER(55)
C****** GET PHONET FILE
        TYPE"*DO YOU WANT TO BANG THE SPECTRAL FILE AGAINST ITSELF?"
        ACCEPT"ENTER (4=YES/3=NO):  ",FSPFSQ
        IF(FSPFSQ.EQ.4)GOTO 37
35      ACCEPT"***ENTER PHONET FILENAME:  "
        READ(11,10)FILE1(1)
36      CALL OPEN(3,FILE1,2,IER)
        CALL CHECK(IER)
        CALL RDBLK(3,0,HEADER,1,IER)
        CALL CHECK(IER)
        IF(OUTD.NE.0)WRITE(OUTD,9001)(HEADER(I5),I5=1,13)
        GOTO 39
C****** GET THE 1ST & LAST PHONETS
37      DO 38 J1=1,13
38      FILE1(J1)=FILE2(J1)
39      TYPE"1ST PHONET TS# IS:",HEADER(55)
        TYPE"LAST PHONET TS# IS: ",HEADER(56)
41      ACCEPT"***ENTER 1ST PHONET TS TO DO:  ",PTS1
        IF(PTS1.GE.HEADER(55))GOTO 42
        TYPE"ERROR: 1ST TIME-SLICE IS TOO SMALL!  MUST BE GREATER"
        TYPE"THAN OR EQUAL TO",HEADER(55)
        GOTO 41
42      ACCEPT"***ENTER LAST PHONET TS TO DO:  ",PTSL
        IF(PTSL.LE.HEADER(56))GOTO 43
        TYPE"ERROR: LAST TIME-SLICE TOO BIG!  MUST BE SMALLER THAN"
        TYPE"OR EQUAL TO",HEADER(56)
        GOTO 42
43      IF(PTS1.LE.PTSL)GOTO 44
        TYPE"ERROR: 1ST TIME-SLICE LARGER THAN LAST!"
        GOTO 41
C****** GET NUMBER OF PHONET TS'S TO BANG AND AMOUNT OF MEMORY
C       THAT WILL TAKE
44      NPTSTB=PTSL-PTS1+1      ;# TS'S TO BANG
        NLDTD=NSTSTB*NPTSTB             ;# DISTANCES TO COMPUTE
        JCTLD=0                         ;SET COUNTER FOR # DISTANCES
        IBP=HEADER(55)
C****** VERIFY # PNTS IN PHONET TIME-SLICES MATCH # PNTS IN SPECTRAL TS'S
        NPPTS=2*HEADER(57)
        IF(NPPTS.EQ.NPSTS)GOTO 60
        TYPE"ERROR: # PNTS IN PHONETIC TIME-SLICE MUST MATCH # PNTS"
        TYPE"IN SPECTRAL TIME-SLICE.  AS SPECIFIED,"
        TYPE"# PNTS IN PTS=",NPPTS
        TYPE"# PNTS IN STS=",NPSTS
        GOTO 35
60      CONTINUE
C****** GET DISTANCE RULE
        ACCEPT"*SELECT DISTANCE RULE (1=M1/2=M2):  ",FDSTR
C****** GET DISTANCE FILE
C****** GET NAME OF FILE TO STASH DISTANCES INTO
65      ACCEPT"***ENTER NAME OF FILE TO RECIEVE DISTANCES:  "
        READ(11,10)FILE3(1)
70      CALL OPEN(2,FILE3,2,IER)
        IF(IER.NE.1)TYPE"ERROR ON OPEN OF DISTANCE FILE, IER= ",IER
        CALL CHKO(IER)
        IF(IER.NE.13)GOTO 75
        TYPE"FILE DOES NOT EXIST.  WILL CREATE IT FOR YOU."
        CALL CFILW(FILE3,2,IER1)
        IF(IER1.NE.1)TYPE"ERROR ON FILE CREATION. IER1= ",IER1
        GOTO 70
75      CONTINUE
C****** STORE FILENAMES IN HEADER OF DISTANCE FILE
        DO 100 J1=1,QTRBLK
100     HEADER(J1)=0.0
        DO 110 J1=1,13
        HEADER(J1)=FILE3(J1)
        J2=J1+13
        HEADER (J2)=FILE2(J1)
        J3=J1+26
110     HEADER(J3)=FILE1(J1)
        HEADER(40)=STS1
        HEADER(41)=STSL
        HEADER(42)=PTS1
        HEADER(43)=PTSL
        HEADER(46)=FSPFSQ        ;SWICH INDICATING IF OBS AND PHONS ARE SAME
        HEADER(48)=NSTSTB        ;# OBS TIME SLICES
        HEADER(49)=NPTSTB        ;# PHON TIME SLICES
        HEADER(50)=NPPTS         ;# WORDS PER TIME-SLICE
C****** DIVVY-UP EXTENDED MEMORY BETWEEN OBSERVATION SPECTRUM,
C       PHONET SPECTRUM, AND THE DISTANCE FILES.
111     NPFS=2+2*IFIX(FLOAT(NSTSTB)*NPSTS/(2*NDP)) ;# MEM BLKS FOR OBS FILE
        NPFP=2+2*IFIX(FLOAT(NPTSTB)*NPPTS/(2*NDP)) ; MEM BLKS FOR PHON FILE
114     CONTINUE
116     J1=IFIX((XMEM-3)/2)*2            ;NEED EVEN # MEM BLKS
        IF(NPFP.LT.J1)NBFPH=NPFP
        IF(NPFP.GE.J1)NBFPH=J1
        J1=IFIX((XMEM-1-NBFPH)/2)*2
        IF(NPFS.LT.J1)NBFOBS=NPFS
        IF(NPFS.GE.J1)NBFOBS=J1
        NBFDIST=1
        IF(NBFOBS.GT.0)GOTO 118
        TYPE"ERROR: INSUFFICIENT EXTENDED MEMORY AVAILABLE."
        TYPE"NEED AT LEAST 14 1K BLOCKS."
        CALL VMEM(XMEM,IER)
        TYPE"YOU HAVE",XMEM,"  1K BLOCKS OF EXTENDED MEMORY AVAILABLE."
        ACCEPT"*HOW MANY DO YOU WANT?  ",XMEM
        XMEM=XMEM-9
        GOTO 114
118     CONTINUE
        HEADER(51)=XMEM
C****** INITIALIZE TO 1ST AVAILABLE EXT MEM BLK
        OBSMAPCNT=9              ;1ST EXT MEM BLK USED FOR OBSERVATIONS
        FONMAPCNT=OBSMAPCNT+NBFOBS             ;1ST EXT MEM BLK
                                               ;USED FOR PHONETS
        DISTMAPCNT=FONMAPCNT+NBFPH             ;1ST EXT MEM BLK
                                               ;USED FOR DISTANCES
        FONBANGCNT=0
        OBSLEFT=NDP
```

FILE:	CHOOS
LANGUAGE:	FORTRAN 5
DATE:	September 21, 1982
AUTHOR:	D. Martin
SUBJECT:	Acoustic Analysis
CALLING SEQUENCE:	CHOOS
DATE OF LAST REVISION:	September 21, 1982

PURPOSE:

This program uses the Array Processor to choose the N-best phonet choices from a file of up to 512 distance measures. The number of choices, N, is selected in the routine which calls this, DSTN.

DESCRIPTION:

Location: DP4:BRATCHET

Size:	CHOOS.FR	3994 bytes
	CHOOS.RB	2870 bytes

PROGRAM USE:

This routine calls CHKJ and is called by DSTN. It uses the Array Processor to choose a selectable number of best distances, best being minimum. It fills the output file with the observation number, the observation energy, the phonet number of the maximum distance, the maximum distance, and consecutive ordered pairs for the choices: the first number being the phonet number, the second is the distance; the first pair is the best choice and the N^{th} pair is the N^{th} choice. Each file segment of 4+2N contains these assignments in the order given here. The output file is an integer file.

Just prior to assignment, the energy value and each distance is checked, while real, for integer overflow. Any value larger than 32767 is assigned the value 32767, which is the largest integer the Eclipse S/250 recognizes. The number of choices is selected in routine DSTN.

A flowchart of this routine is in Figure 75. When called, the distances are loaded into Array Processor memory. After the energy and observation number are loaded into the output array DIST, the Array Processor is used to first get the input distance file maximum and corresponding phonet number. Then the routine enters a loop which successively gets the minimum distance and phonet number, replaces the minimum distance with the max, then gets the next minimum. This continues until the selected number of choices is obtained.

LIST OF VARIABLES:

VARIABLE	TYPE	PURPOSE
NPFSR	Integer	Energy in observation.
NPTSTB	Integer	Number of phonets.
HOLDIST	Real Array	Holds distances.
WKOBS	Real Array	Holds distances in APM.
JOUTDIST	Integer	Counts output array elements.
OBSBANGCNT	Integer	Counts observations done.

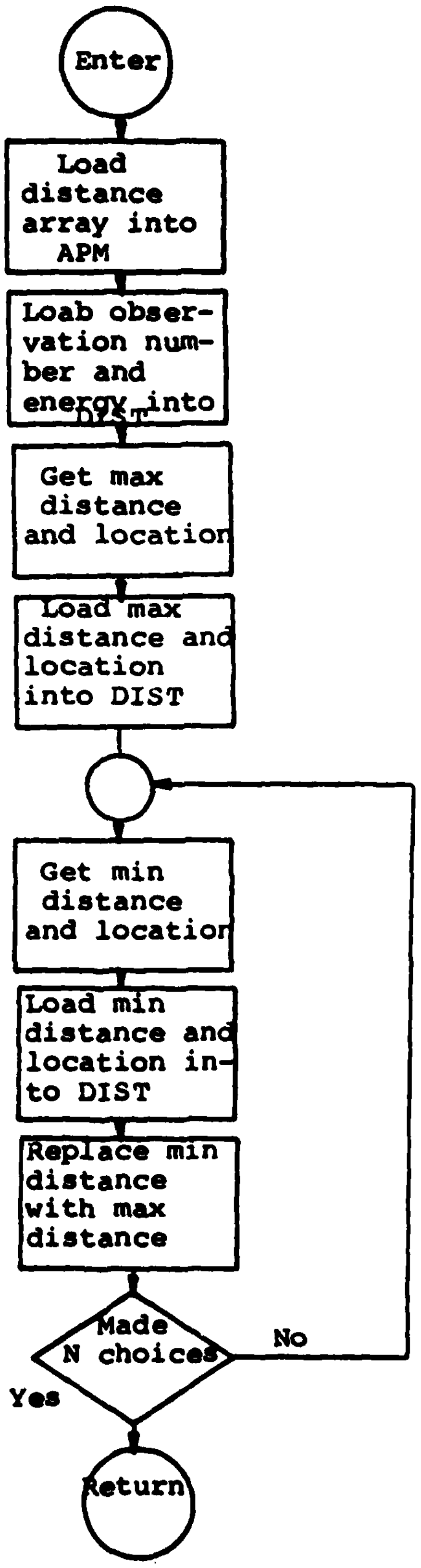

Enter
Load distance array into APM
Loab observation number and energy into DIST
Get max distance and location
Load max distance and location into DIST
Get min distance and location
Load min distance and location into DIST
Replace min distance with max distance
Made N choices
No
Yes
Return

FIGURE 75. Flowchart of CHOOS

VARIABLE	TYPE	PURPOSE
DIST	Integer	Output array.
MAXDIST	Integer	Maximum distance in HOLDIST.
MAXDISTLOC	Integer	Phonet number corresponding to MAXDIST.
MINDIST	Integer	Minimum distance in HOLDIST.
MINDISTLOC	Integer	Phonet number corresponding to MINDIST.

SWITCH SETTINGS: None.

RELATED PROGRAMS:

PLTO, PLTA, PLTS, PLTN, PLTT, GRPH2, DRVR, DSTA, DSTN, S128, S64, CHKO, CHKJ, FCTR, GFDB.

```
C****** ROUTINE CHOOS.  THIS ROUTINE IS FOR FORTRAN 5 ! !
C       THIS ROUTINE USES THE ARRAY PROCESSOR TO CHOOSE THE N-BEST PHONETS
C       FROM AN ARRAY OF UP TO 512 DISTANCE MEASURES.  THE NUMBER OF
C       CHOICES, N, IS SELECTED IN THE ROUTINE WHICH CALLS THIS ONE, DSTN.
C       BY:  CAPT DAN MARTIN
C       DATE:  9/21/82
C       SUBJ:  ACOUSTIC ANALYSIS
C       THIS ROUTINE CALLS CHKJ AND IS CALLED BY DSTN.  IT FILLS AN OUTPUT
C       FILE WITH THE OBSERVATION NUMBER, THE OBSERVATION ENERGY, THE
C       PHONET NUMBER OF THE MAXIMUM DISTANCE, THE MAXIMIMUM DISTANCE,
C       AND CONSECUTIVE ORDERED PAIRS FOR EACH CHOICE: (PHONET NUMBER,
C       NEXT SMALLEST DISTANCE).  THE OUTPUT FILE CONSISTS OF A HEADER
C       OF ONE DISK BLOCK AND CONTIGUOUS UNITS OF 4+2N POINTS IN LENGTH,
C       N BEING THE NUMBER OF CHOICES MADE.  EACH UNIT CONTAINS THE ABOVE
C       LISTED VALUES FOR EACH OBSERVATION.
C       THIS ROUTINE ACCOMPLISHES THE FOLLOWING TASKS:
C               1)  LOAD DISTANCE ARRAY INTO ARRAY PROCESSOR MEMORY
C               2)  LOAD OBSERVATION NUMBER AND ENERGY INTO OUTPUT
C               ARRAY DIST
C               3)  GET MAX DISTANCE AND LOCATION
C               4)  LOAD MAX DISTANCE AND LOCATION INTO DIST
C               5)  GET MIN DISTANCE AND LOCATION
C               6)  LOAD MIN DISTANCE AND LOCATION INTO DIST
C               7)  REPLACE MIN DISTANCE WITH MAX DISTANCE
C               8)  GO TO TASK 5 UNTIL ALL CHOICES ARE MADE
C       FOR MORE INFORMATION SEE THE USERS MANUAL OR MY THESIS.
        SUBROUTINE CHOOS
        OVERLAY OCHOO
        INCLUDE"ARRAYP:F5APS.FR"
        PARAMETER NDP=1024
        PARAMETER NDPM1=1023
        PARAMETER QTRBLK=256
        REAL WKFON,WKOBS,NPFSR,NPFPR,FACTOR,FON,OBS,HOLDIST
        REAL MAXDIST,MINDIST
        INTEGER FSPFN,FILE2,HEADER,STS1,STSL,NSTSTB,NPSTS,NPFS
        INTEGER DIST,FSPFSQ,FILE1,PTS1,PTSL,JOUTDIST
        INTEGER NPTSTB,NPPTS,NPFP,FBSTR,FILE3,OUTD,FLAG,CB1,SKIP
        INTEGER XMEM,NBFDIST,NBFOBS,NBFPH,CNTOBS,CNTFON,OBSMAPCNT
        INTEGER FONMAPCNT,FONBANGCNT,OBSLEFT,NQBFOBS,NQBFPH,NQBFDIST
        INTEGER DISTMAPCNT,OBSBANGCNT,FONLEFT,ICNTOBS,IFBC,GSPCT
        INTEGER NPLTB,JFONMAPCNT,JOBSMAPCNT,DISTMAPCNTMAX,NDBFDIST
        INTEGER JDISTNBR,JCNTOBS,JCNTFON,ICNTFON,NFONLD,INITMEM
        INTEGER MAXDISTLOC,MINDISTLOC,NCHOICES,NBRPHONES

        COMMON / APM / WKOBS(NDP),WKFON(NDP)
        COMMON / VALS / OBS(NDP),FON(NDP),DIST(NDP),OBSLEFT,FSPFN
        COMMON / VALS / INITMEM,FILE3(13),HEADER(QTRBLK),NPFSR,NPFPR
        COMMON / VALS / PTS1,PTSL,NPTSTB,NPPTS,NPFP,FBSTR,OUTD,NBFDIST
        COMMON / VALS / NBFOBS,NBFPH,CNTOBS,CNTFON,OBSMAPCNT,FONMAPCNT
        COMMON / VALS / FONBANGCNT,NQBFOBS,NQBFPH,NQBFDIST,DISTMAPCNT
        COMMON / VALS / ICNTOBS,ICNTFON,IFBC,JFONMAPCNT,NFONLD
        COMMON / VALS / OBSBANGCNT,FONLEFT,NPLTB,JOBSMAPCNT,NDBFDIST
        COMMON / VALS / DISTMAPCNTMAX,JOUTDIST,JDISTNBR,JCNTOBS,JCNTFON
        COMMON / VALT / SKIP,STSL,NSTSTB,NPSTS,NPFS,XMEM,FACTOR,FSPFSQ
        COMMON / VALT / FILE1(13),FILE2(13),FLAG,STS1
        COMMON / VALV / CB1(0:CBMAX)
        COMMON / VALU / HOLDIST(NDP),MAXDIST,MINDIST,MAXDISTLOC,MINDISTLOC
        COMMON / VALU / NCHOICES,NBRPHONES

        SKIP=0
C****** INITIALIZE APS AND APM MAP FOR ERDB
        IF(SKIP.EQ.0)GOTO 502
        SKIP=0
        CALL APINIT(NIL,DTARAY,9,INITMEM,IER)
        CALL APMAP(DTARAY,0,4,IER)
502     CONTINUE
C****** LOAD ARRAY HOLDIST INTO WKOBS
        CALL APSETL(NPTSTB,IER)
        CALL CBSET(CB1,CBAAMM,HOLDIST,CBAXR,WKOBS,IER)
        CALL VLDR(CB1)
C****** LOAD OBSERVATION # INTO OUTPUT ARRAY DIST
        DIST(JOUTDIST)=OBSBANGCNT-1
        CALL CHKJ                          ;SEE IF DIST IS FULL.
        JOUTDIST=JOUTDIST+1
C****** LOAD OBSERVATION ENERGY NEXT
        NPFSR=NPFSR/100           ;SCALE ENERGY
        IF(NPFSR.GT.32767)NPFST=32767
        DIST(JOUTDIST)=NPFSR
        CALL CHKJ
        JOUTDIST=JOUTDIST+1
C****** GET MAX DISTANCE AND ITS LOCATION
        CALL CBSET(CB1,CBAXR,WKOBS,CBAAMM,MAXDIST,CBAIMM,MAXDISTLOC,IER)
        CALL VMXR(CB1)
        MAXDISTLOC=MAXDISTLOC+1
        DIST(JOUTDIST)=MAXDISTLOC
        CALL CHKJ
        JOUTDIST=JOUTDIST+1
        DIST(JOUTDIST)=MAXDIST
        CALL CHKJ
        JOUTDIST=JOUTDIST+1
C****** GET MINIMUM DISTANCES AND THEIR LOCATIONS, I.E., PHONET #S
        DO 2000 J2=1,NCHOICES
        CALL CBSET(CB1,CBAXR,WKOBS,CBAAMM,MINDIST,CBAIMM,MINDISTLOC,IER)
        CALL VMNR(CB1)                                ;GET MINIMUM
        DIST(JOUTDIST)=MINDISTLOC+1                   ;LOCATION OF MINIMUM
        WKOBS(DIST(JOUTDIST))=MAXDIST
        CALL CHKJ
        JOUTDIST=JOUTDIST+1
        DIST(JOUTDIST)=MINDIST
        CALL CHKJ
2000    JOUTDIST=JOUTDIST+1

        RETURN
        END
```

FILE:	FCTR
LANGUAGE:	FORTRAN 5
AUTHOR:	D. Martin
DATE:	September 21, 1982
SUBJECT:	Acoustic Analysis
CALLING SEQUENCE:	FCTR (W, M, H, Q, A, B, C, D)
DATE OF LAST REVISION:	September 21, 1982

PURPOSE:

This subroutine is called by GFDB and calls no subroutines. It factors an integer into a linear combination of four terms. GFDB uses it to determine the unit (disk block, etc.) and starting data point in that unit at which specified data reside. For example, if the N^{th} element of an integer disk file is to be accessed, then GFDB will call FCTR to specify the disk block and location in that block where that N^{th} element resides.

DESCRIPTION:

Location: DP4:BRATCHET

Size:	FCTR.FR	605 bytes
	FCTR.RB	218 bytes

ARGUMENT STRUCTURE:

ARGUMENTS	TYPE	PURPOSE
W	Integer	Integer to be factored.
M	Integer	Factor.
A	Integer	Coefficient of M.
H	Integer	Factor.

ARGUMENTS	TYPE	PURPOSE
B	Integer	Coefficient of H.
Q	Integer	Factor.
C	Integer	Coefficient of Q.
D	Integer	Coefficient of one.

PROGRAM USE:

Values of W, M, H, and Q are passed to FCTR. FCTR returns values of A, B, C, and D for which:

$$W = AM + BH + CQ + D.$$

RELATED PROGRAMS:

PLTO, PLTS, PLTA, PLTN, PLTT, DRVR, DRSQ, DSTA, DSTN, S64, S128, GFDB, GRPH2, CHKO, CHKJ.

```
C****** ROUTINE FCTR.  THIS ROUTINE FOR FORTRAN 5 ! !
C       THIS SUBROUTINE IS CALLED BY CFDB AND CALLS NO SUBROUTINES.  IT
C       FACTORS AN INTEGER INTU A LINEAR COMBINATION OF FOUR 1-ST ORDER
C       TERMS.  CFDB USES IT TO DETERMINE THE DISK BLOCK AND STARTING
C       ELEMENT IN A DISK FILE GIVEN A SPECIFIED TIME SLICE OR SPECTRAL
C       SLICE NUMBER.
C       BY:  CAPT DAN MARTIN
C       DATE:  9/21/82
C       SUBJ:  ACOUSTIC ANALYSIS
C       SEE THE USERS MANUAL OR MY THESIS FOR MORE INFORMATION.
        SUBROUTINE FCTR(W,M,H,Q,A,B,C,D)
        INTEGER W,M,H,Q,A,B,C,D,X,Y
C****** W=A*M+B*H+C*Q+D
        A=W/M
        X=W-A*M
        B=X/H
        Y=X-B*H
        C=Y/Q
        D=Y-C*Q
        RETURN
        END
```

FILE:	GFDB
LANGUAGE:	FORTRAN 5
DATE:	September 21, 1982
AUTHOR:	D. Martin
SUBJECT:	Acoustic Analysis
CALLING SEQUENCE:	GFDB (IA, IB)
DATE OF LAST REVISION:	September 21, 1982

PURPOSE:

This routine is called by PLTA, PLTS, PLTN, PLTT, DSTA, and DSTN. It calls FCTR to compute the location of a specified data point.

DESCRIPTION:

Location: DP4:BRATCHET

Size:	GFDB.FR	644 bytes
	GFDB.RB	278 bytes

ARGUMENT STRUCTURE:

ARGUMENT	TYPE	PURPOSE
IA	Integer	Passes file slice number. Returns first disk block.
IB	Integer	Passes number of integer words per slice. Returns first data word.

PROGRAM USE:

The plot routines PLTA, PLTS, PLTN, PLTT, DSTA, and DSTN use this routine to locate specified data.

```
C****** ROUTINE GFDB.  THIS ROUTINE IS FOR FORTRAN 5 ! !
C       THIS ROUTINE GETS THE 1ST DISK BLOCK (256 INTEGER
C       WORDS) AND THE 1ST ELEMENT OF THE ARRAY WHICH WAS LOADED
C       AT THAT DISK BLOCK.  THE FIRST ELEMENT OF THE ARRAY
C       IS NUMBER ONE.
C       BY:  CAPT DAN MARTIN
C       DATE:  9/21/82
C       SUBJ:  ACOUSTIC ANALYSIS
C       WHEN PASSED: IA=FILE SLICE NUMBER
C                    IB=NUMBER OF INTEGER WORDS PER SLICE
C       WHEN RETURNED: IA=1ST DISK BLOCK
C                      IB=1ST DATA WORD
C       FOR MORE INFORMATION, SEE THE USERS MANUAL OR MY THESIS.

        SUBROUTINE GFDB(IA,IB)
        INTEGER W,A,B,C,D

        W=IB*(IA-1)
        CALL FCTR(W,1024,512,256,A,B,C,D)
        IA=4*A+2*B+C
        IB=D+1
        RETURN
        END
```

FILE:	CHKJ
LANGUAGE:	FORTRAN 5
DATE:	September 21, 1982
AUTHOR:	D. Martin
SUBJECT:	Acoustic Analysis
CALLING SEQUENCE:	CHKJ
DATE OF LAST REVISION:	September 21, 1982

PURPOSE:

This routine is called by DSTN and calls no subroutines. It checks the number of distances accumulated, writes them if 1024 distances are accumulated, resets the counter, and increments a pointer to the next disk block to write.

DESCRIPTION:

Location: DP4:BRATCHET

Size:	CHKJ.FR	2324 bytes
	CHKJ.RB	1012 bytes

LIST OF VARIABLES:

VARIABLE	TYPE	PURPOSE
JOUTDIST	Integer	Number of distances accumulated.
DIST	Integer Array	Holds distances accumulated.
NDBFDIST	Integer	Disk block to write output to.

PROGRAM USE:

This routine is designed for use by DSTN, to be called each time JOUTDIST is incremented.

__RELATED PROGRAMS__:

PLTO, PLTA, PLTS, PLTN, PLTT, FCTR, GRPH2, CHOOS, DRVR, DRSQ, S128, S64, DSTA, DSTN, CKHJ, CHKO.

```
C****** ROUTINE CHKJ.  THIS ROUTINE FOR FORTRAN 5 ! !
C       THIS ROUTINE IS CALLED BY DSTN AND CALLS NO SUBROUTINES.  IT
C       CHECKS THE NUMBER OF DISTANCES ACCUMULATED, WRITES THEM IF
C       1024 HAVE BEEN ACCUMULATED, RESETS THE COUNTER, AND INCREMENTS
C       A POINTER TO THE NEXT DISK BLOCK TO WRITE TO.
C       BY:  CAPT DAN MARTIN
C       DATE:  9/21/82
C       SUBJ:  ACOUSTIC ANALYSIS
C       THIS ROUTINE IS DESIGNED FOR USE BY DSTN TO BE CALLED EACH TIME
C       JOUTDIST IS INCREMENTED.  FOR MORE INFORMATION, SEE THE
C       USERS MANUAL FOR DSTN OR THIS ROUTINE, OR SEE MY THESIS.


        SUBROUTINE CHKJ
        OVERLAY OCHKJ
        PARAMETER NDP=1024
        PARAMETER NDPM1=1023
        PARAMETER QTRBLK=256
        REAL WKFON,WKOBS,NPFSR,NPFPR,FACTOR,FON,OBS,HOLDIST
        REAL MAXDIST,MINDIST
        INTEGER FSPFN,FILE2,HEADER,STS1,STSL,NSTSTB,NPSTS,NPFS
        INTEGER DIST,FSPFSQ,FILE1,PTS1,PTSL,JOUTDIST
        INTEGER NPTSTB,NPPTS,NPFP,FDSTR,FILE3,OUTD,FLAG,CB1,SKIP
        INTEGER XMEM,NBFDIST,NBFOBS,NBFPH,CNTOBS,CNTFON,OBSMAPCNT
        INTEGER FONMAPCNT,FONBANGCNT,OBSLEFT,NQBFOBS,NQBFPH,NQBFDIST
        INTEGER DISTMAPCNT,OBSBANGCNT,FONLEFT,ICNTOBS,IFBC,GSPCT
        INTEGER NPLTB,JFONMAPCNT,JOBSMAPCNT,DISTMAPCNTMAX,NDBFDIST
        INTEGER JDISTNBR,JCNTOBS,JCNTFON,ICNTFON,NFONLD,INITMEM
        INTEGER MAXDISTLOC,MINDISTLOC,NCHOICES,NBRPHONES

        COMMON / APM / WKOBS(NDP),WKFON(NDP)
        COMMON / VALS / OBS(NDP),FON(NDP),DIST(NDP),OBSLEFT,FSPFN
        COMMON / VALS / INITMEM,FILE3(13),HEADER(QTRBLK),NPFSR,NPFPR
        COMMON / VALS / PTS1,PTSL,NPTSTB,NPPTS,NPFP,FDSTR,OUTD,NBFDIST
        COMMON / VALS / NBFOBS,NBFPH,CNTOBS,CNTFON,OBSMAPCNT,FONMAPCNT
        COMMON / VALS / FONBANGCNT,NQBFOBS,NQBFPH,NQBFDIST,DISTMAPCNT
        COMMON / VALS / ICNTOBS,ICNTFON,IFBC,JFONMAPCNT,NFONLD
        COMMON / VALS / OBSBANGCNT,FONLEFT,NPLTB,JOBSMAPCNT,NDBFDIST
        COMMON / VALS / DISTMAPCNTMAX,JOUTDIST,JDISTNBR,JCNTOBS,JCNTFON
        COMMON / VALT / SKIP,STSL,NSTSTB,NPSTS,NPFS,XMEM,FACTOR,FSPFSQ
        COMMON / VALT / FILE1(13),FILE2(13),FLAG,STS1
        COMMON / VALU / HOLDIST(NDP),MAXDIST,MINDIST,MAXDISTLOC,MINDISTLOC
        COMMON / VALU / NCHOICES,NBRPHONES

        IF(JOUTDIST.LE.NDP)GOTO 365     ;DIST ARRAY NOT FULL YET
        JOUTDIST=1                      ;NEED WRITE TO DISK IF FULL
        IF(OUTD.NE.0)WRITE(OUTD,9000)(DIST(J2),J2=1,NDP)
C****** WRITE 4 QTR BLKS OF DISTANCES
        CALL EWRB(2,NDBFDIST,36+NQBFOBS+NQBFPH,4,IER)
        IF(IER.NE.1)TYPE"ERROR ON EWRB OF DISTANCES! IER=",IER
        NDBFDIST=NDBFDIST+4
365     CONTINUE
9000    FORMAT(5G12.4)
        RETURN
```

END

FILE:	PLTO
LANGUAGE:	FORTRAN 5
DATE:	September 21, 1982
AUTHOR:	D. Martin
SUBJECT:	Graphics
CALLING SEQUENCE:	PLTO
DATE OF LAST REVISION:	September 21, 1982

PURPOSE:

This routine calls one of four plot routines selected from a displayed menu. The graph is displayed on a Tektronix 4010-1 Graphics Terminal using the routine GRPH2 by G. Shaw as modified by L. Kizer and D. Zambon.

DESCRIPTION:

Location: DP4:BRATCHET

Size:	PLTO.FR	1491 bytes
	PLTO.RB	1650 bytes

PROGRAM USE:

This program is called by DRVR and calls PLTS, PLTA, PLTN, and PLTT. When called, this routine presents a menu from which the operator is to choose one of five options, one being to return to DRVR. If the option to plot spectrum is chosen, PLTS is called to plot the spectrum of a time slice from a disk file computed by DRSQ. If the option to plot all distances is selected, PLTA is called to plot distances between one observation and selected phonets from a disk file computed by DSTA. If the option to plot N-best distances is selected, PLTN is called to plot observation energy and selected distances from a disk file computed by DSTN. Finally, if the option to plot an integer file is selected, then one unit of up to 512 elements from any integer disk file is plotted.

LIST OF VARIABLES:

VARIABLE	TYPE	PURPOSE
OUTD	Integer	Currently unused, but can be used as unit number for screen output.
SKIP	Integer	Switch to select plot option.

SWITCH SETTINGS:

SWITCH		SETTINGS
SKIP	=	6 to call PLTS to plot spectrum
	=	7 to call PLTA to plot distances
	=	8 to call PLTN to plot N-best distances, worst distance, and observation energy
	=	9 to call PLTT to plot a segment of an integer file
	=	10 to return to DRVR

RELATED PROGRAMS:

PLTA, PLTS, PLTN, PLTT, DRVR, DSTA, DSTN, DRSQ, CHKJ, CHKO, CHOOS, FCTR, GFDB, S128, S64, GRPH2.

```
C****** ROUTINE PLTO.  THIS ROUTINE IS FOR FORTRAN 5 ! !
C       THIS ROUTINE CALLS PLTA, PLTS, PLTN, AND PLTT , AND IS CALLED
C       BY DRVR.  IT SELECTS THE APPROPRIATE PLOT ROUTINE TO PLOT THE
C       DATA AS REQUESTED.
C       BY:  CAPT DAN MARTIN
C       DATE:  9/21/82
C       SUBJ:  ACOUSTIC ANALYSIS
C       FOR MORE INFORMATION SEE THE USERS MANUAL OR MY THESIS.
                SUBROUTINE PLTO
        OVERLAY OPLTO
        INCLUDE "ARRAYP:F5APS.FR"
        PARAMETER HNDP=1024
        PARAMETER QTRBLK=256
        REAL DATA,XMIN,XMAX
        INTEGER CSPCT,FILE2,HEADER,FSPFN,OUTD,OBSNBR,PHON1,PHONL
        INTEGER DIST,FONNBR,FRSTDP,FRSTDBLK,ICNT,D1,IER,SKIP

        COMMON / VALS / DATA(HNDP),DIST(1024),OBSNBR,FSPFN,OUTD,PHON1
        COMMON / VALS / PHONL,ICNT,IER,FONNBR,FRSTDP,FRSTDBLK,FILE2(13)
        COMMON / VALS / HEADER(QTRBLK),SKIP
        COMMON / VALT / D1(37),ISXMEM

        TYPE"****YOU ARE NOW IN ROUTINE PLTO.FR****"
        OUTD=10

40      TYPE"**WHAT WILL YOU HAVE ME PLOT?"
        TYPE"****CHOOSE OPTION****"
        TYPE"*"
        TYPE"6 TO PLOT SPECTRUM."
        TYPE"7 TO PLOT ALL DISTANCES."
        TYPE"8 TO PLOT N-BEST DISTANCES."
        TYPE"9 TO PLOT INTEGER FILE."
        TYPE"10 TO RETURN TO THE PROGRAM WHICH CALLED THIS ONE."
        ACCEPT"**ENTER DESIRED OPTION:  ",SKIP

100     IF(SKIP.NE.6)GOTO 110
        CALL PLTS
        GO TO 40
110     IF(SKIP.NE.7)GOTO 120
        CALL PLTA
        GOTO 40
120     IF(SKIP.NE.8)GOTO 130
        CALL PLTN
        GOTO 40
130     IF(SKIP.NE.9)GOTO 140
        CALL PLTT
        GOTO 40
140     IF(SKIP.EQ.10)GOTO 1000

        TYPE"ERROR: COULDN'T FIND YOUR OPTION."
        TYPE"YOU SELECTED OPTION:  ",SKIP
        GOTO 40
1000    CALL RESET
        CALL OVEXIT(OPLTO,IER)
```

```
      CALL CHECK(IER)
      RETURN
      END
```

FILE:	PLTA
LANGUAGE:	FORTRAN 5
DATE:	September 21, 1982
AUTHOR:	D. Martin
SUBJECT:	Graphics
CALLING SEQUENCE:	PLTA
DATE OF LAST REVISION:	September 21, 1982

PURPOSE:

This routine plots distances from a disk file computed by DSTA. Distances between an observation and all phonets specified by the operator are displayed on a Tektronix 4010-1 Graphics terminal using the plot routine GRPH2 by G. Shaw as modified by L. Kizer and D. Zambon.

DESCRIPTION:

Location: DP4:BRATCHET

Size:	PLTA.FR	4371 bytes
	PLTA.RB	6712 bytes

PROGRAM USE:

This program is called by PLTO and calls GFDB and GRPH2. It plots distances from a disk file between a specified observation and the first and last phonets specified.

The operator is first prompted for an option to print numerical values of data to be plotted. The data values plotted can be displayed on the operator's terminal, written to the line printer, or not displayed at all. Next, the operator is prompted for the name of the disk file holding the data to be plotted. The file header is read and identifying information from it displayed for the operator. Then the desired observation number, and the numbers of the first and last phonet to be plotted, are requested. When signalled to proceed, the program will display the plot.

<u>LIST OF VARIABLES</u>:

<u>VARIABLE</u>	<u>TYPE</u>	<u>PURPOSE</u>
OUT	Integer	Switch for data output option.
OUTD	Integer	Channel number.
FILE3	Integer Array	Holds disk file name.
OBSNBR	Integer	Observation number.
PHON1	Integer	First phonet.
PHONL	Integer	Last phonet.
FONNBR	Integer	Number of distances to plot.
FRSTDBLK	Integer	First disk block to read.
FRSTDP	Integer	First data point in disk block.
DATA	Real Array	Holds data to plot.
DIST	Integer Array	Data read from disk.

<u>SWITCH SETTINGS</u>:

<u>SWITCH</u>		<u>SETTING</u>
OUT	=	10 for screen output of plotted data
	=	12 for line printer output
	=	0 for no output of numbers.

RELATED PROGRAMS:

PLTO, PLTS, PLTN, PLTT, DRSQ, S128, S64, DSTA, DSTN, CHOOS, CHKJ, CHKO, FCTR, GFDB, DRVR, GRPH2.

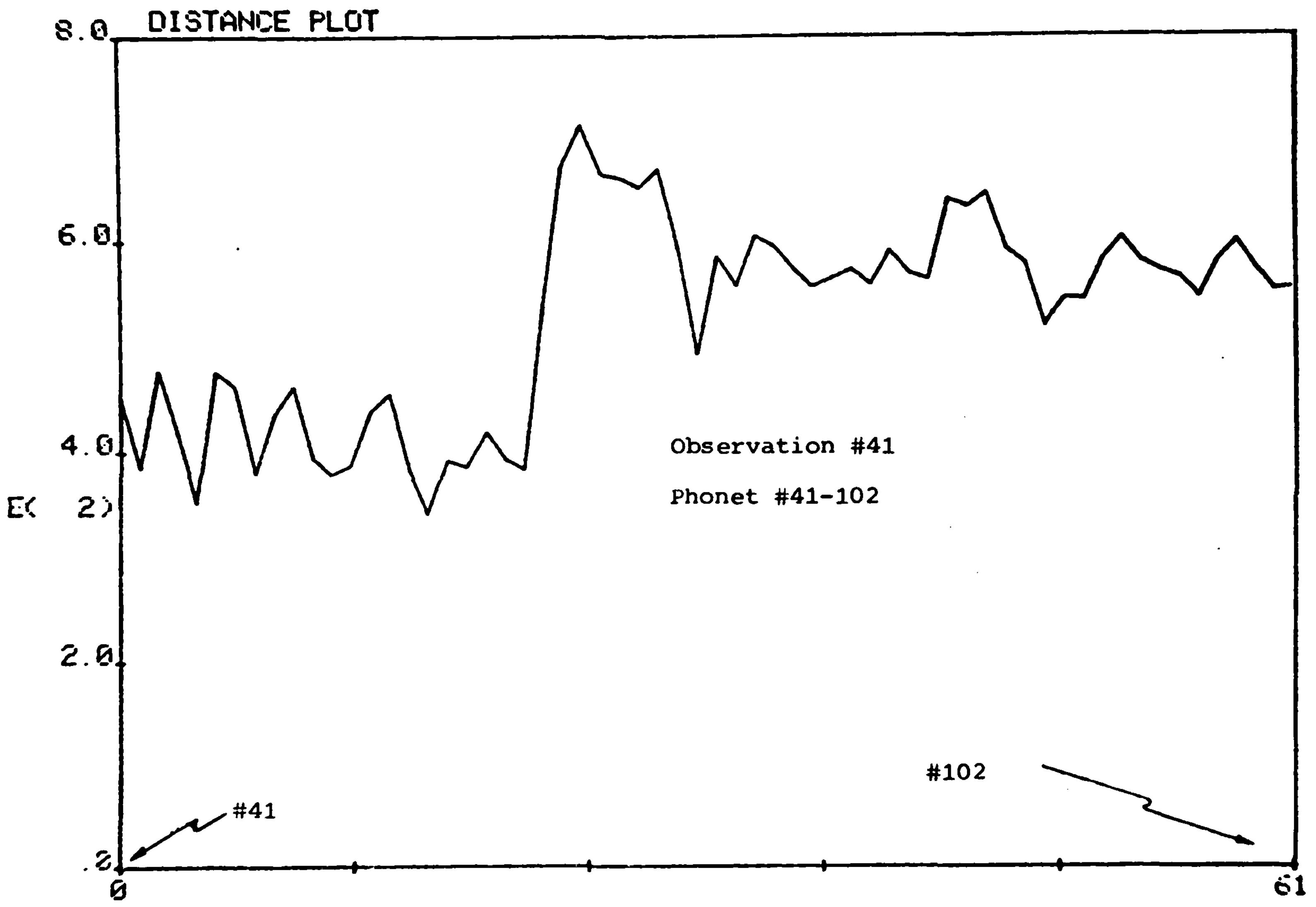


FIGURE 76. Example of PLTA

```
C****** ROUTINE PLTA.  THIS ROUTINE IS FOR FORTRAN 5 ! !
C       THIS ROUTINE IS CALLED BY PLTO AND CALLS THE PLOT ROUTINES GRPH2
C       AND GFBD.  GRPH2 WAS WRITTEN BY G.SHAW AND MODIFIED BY L.KIZER
C       AND D.ZAMBON.  IT IS IN THE SPEECH LAB PROGRAM LIBRARY.
C       THIS ROUTINE PLOTS DISTANCES COMPUTED BY DSTA.  IT WILL PLOT
C       DISTANCES BETWEEN A SPECIFIED OBSERVATION AND SPECIFIED PHONETS.
C       THE USER IS PROMPTED FOR THIS INFORMATION AS WELL AS FOR THE
C       FILE NAME.
C       BY:  CAPT DAN MARTIN
C       DATE:  9/21/82
C       SUBJ:  ACOUSTIC ANALYSIS
C       PLTA ACCOMPLISHES THE FOLLOWING TASKS:
C               1)  GET DISTANCE FILE
C               2)  DISPLAY INFORMATION FROM HEADER TO IDENTIFY THE FILE
C               3)  GET OBSERVATION NUMBER AND PHONET NUMBERS
C               4)  LOCATE DATA IN DISK FILE
C               5)  PREPARE DATA FOR PLOT ROUTINE GRPH2
C               6)  CALL THE PLOT ROUTINE
C       FOR MORE INFORMATION SEE THE USERS MANUAL OR MY THESIS.

        SUBROUTINE PLTA
        OVERLAY OPLTA
        INCLUDE"ARRAYP:F5APS.FR"
        PARAMETER HNDP=512
        PARAMETER QTRBLK=256
        REAL DATA,XMIN,XMAX
        INTEGER GSPCT,FILE3,HEADER,FSPFN,OUTD,OBSNBR,PHON1,PHONL
        INTEGER DIST,FONNBR,FRSTDP,FRSTDBLK,ICNT,D1,IER,OUT

        COMMON / VALS / DATA(HNDP),DIST(1024),OBSNBR,FSPFN,OUTD,PHON1
        COMMON / VALS / PHONL,ICNT,IER,FONNBR,FRSTDP,FRSTDBLK,FILE3(13)
        COMMON / VALS / HEADER(QTRBLK)
        COMMON / VALT / D1(37)

        TYPE"****YOU ARE NOW IN ROUTINE PLTD.FR****"
30      TYPE"DO YOU WANT OUTPUT OF DATA PLOTTED?"
        TYPE"ENTER:   10 FOR SCREEN OUPUT OF NUMBERS"
        TYPE"         12 FOR PRINTED NUMERICAL OUTPUT"
        ACCEPT"            0 FOR NO NUMERICAL OUTPUT:  ",OUT
        IF(OUT.EQ.0.OR.OUT.EQ.10.OR.OUT.EQ.12)GOTO 40
        TYPE"ERROR:  ILLEGAL INPUT."
        GOTO 30
40      OUTD=10

C****** GET DISTANCE FILE NAME
50      ACCEPT"**ENTER NAME OF FILE HOLDING DISTANCES: "
        READ(11,100)FILE3(1)
100     FORMAT(S13)
120     CALL CLOSE(2,IER)
        CALL OPEN(2,FILE3,2,IER)
        CALL CHECK(IER)
        IF(IER.NE.1)TYPE"ERROR ON OPEN OF SPECTRAL FILE, IER= ",IER
        GOTO 180
150     CONTINUE
```

```
C****** GET NUMBERS OF 1ST & LAST TIME-SLICES OF SPECTRUM
180     CALL RDBLK(2,0,HEADER,1,IER)
        CALL CHECK(IER)
        WRITE(OUTD,9110)
        WRITE(OUTD,9100)(HEADER(I5),I5=1,65)
        WRITE(OUTD,9000)(HEADER(I5),I5=1,13)
        WRITE(OUTD,9001)(HEADER(I5),I5=14,26)
        WRITE(OUTD,9004)HEADER(40),HEADER(41)
        WRITE(OUTD,9002)(HEADER(I5),I5=27,39)
        WRITE(OUTD,9005)HEADER(42),HEADER(43)
        ACCEPT"IS THIS THE FILE YOU'RE AFTER? (1=YES/0=NO): ",FSPFN
        IF(FSPFN.EQ.0)GOTO 50
190     ACCEPT"ENTER THE OBSERVATION NUMBER:  ",OBSNBR
        IF(OBSNBR.GE.HEADER(40).AND.OBSNBR.LE.HEADER(41))GOTO 200
        TYPE"ERROR:  THIS NUMBER MUST LIE IN THE RANGE"
        TYPE"[",HEADER(40),",",HEADER(41),"]"
        GOTO 190
200     ACCEPT"ENTER THE FIRST PHONET NUMBER:  ",PHON1
        IF(PHON1.GE.HEADER(42).AND.PHON1.LE.HEADER(43))GOTO 220
        TYPE"ERROR:  THIS NUMBER MUST LIE IN THE RANGE"
        TYPE "[",HEADER(42),",",HEADER(43),"]"
        GOTO 200
220     ACCEPT"ENTER THE LAST PHONET NUMBER:  ",PHONL
        IF(PHONL.GE.PHON1.AND.PHONL.LE.HEADER(43))GOTO 240
        TYPE"ERROR:  THIS NUMBER MUST BE IN THE RANGE"
        TYPE"[",PHON1,",",HEADER(43),"]"
        GOTO 200
240     FONNBR=PHONL-PHON1+1
        IF(FONNBR.LE.HNDP)GOTO 260
        TYPE"ERROR:  YOU REQUESTED MORE THAN 512 DISTANCES."
        GOTO 200
260     CONTINUE
        FONNBR=HEADER(43)-HEADER(42)+1
C****** GET DISK BLOCK AND 1ST DATA POINT
        IA=OBSNBR-HEADER(40)+1
        IB=FONNBR
        CALL GFDB(IA,IB)
        FRSTDBLK=IA+1
        FRSTDP=IB
        CALL RDBLK(2,FRSTDBLK,DIST,3,ICNT,IER)
        IF(IER.NE.9)GOTO 300
        TYPE"READ EOF!  SUCCESSFULLY TRANSFERRED",ICNT,"  QTR BLKS"
        TYPE"PROCEEDING WITH  ",ICNT,"QTR BLOCKS TRANSFERRED."
        IER=1
300     CALL CHECK(IER)
        CALL CLOSE(2,IER)
        CALL CHECK(IER)

        IF(OUT.EQ.0)GOTO 310
        IF(OUT.EQ.12)OUTD=12
C****** MOVE THE DATA UP-FRONT AND INTO THE ARRAY TO PLOT
310     IL=PHON1-HEADER(42)+1
        IU=PHONL-HEADER(42)+1
        FRSTDP=FRSTDP+IL-1
        DO 320 J1=1,IU-IL+1
320     DATA(J1)=DIST(J1+FRSTDP-1)
        IF(OUT.EQ.0)GOTO 322
        WRITE(OUTD,9100)(DATA(I5),I5=1,IU-IL+1)
322     ACCEPT"OK TO CONTINUE? (ENTER ANY NUMBER)  ",I5
        IF(I5.NE.0)GOTO 324
324     CONTINUE
        FONNBR=PHONL-PHON1+1 .
        CALL GRPH2("  DISTANCE PLOT",1,DATA,U,FONNBR,0,XMIN,XMAX,0)
        ACCEPT"OK TO CONTINUE? (ENTER ANY NUMBER)  ",I5
        IF(I5.EQ.0)GOTO 350
350     CONTINUE
9100    FORMAT(5G12.4)
9110    FORMAT(T5,"HERE'S THE FIRST 65 ELEMENTS OF THE HEADER:")
9000    FORMAT(T3,"DISTANCE FILE NAME IS:  ",13S2)
9001    FORMAT(T5,"OBSERVATION FILE NAME WAS:  ",13S2)
9002    FORMAT(T5,"PHONET FILE NAME WAS:  ",13S2)
9004    FORMAT(T2,"FIRST OBSERVATION TS# =  ",I5,"   AND LAST =  ",I6)
9005    FORMAT(T2,"FIRST PHONET TS# =  ",I5,"   AND LAST =  ",I6)

        RETURN
        END
```

FILE:	PLTS
LANGUAGE:	FORTRAN 5
DATE:	September 21, 1982
AUTHOR:	D. Martin
SUBJECT:	Graphics
CALLING SEQUENCE:	PLTS
DATE OF LAST REVISION:	September 21, 1982

PURPOSE:

This routine plots spectrum from a disk file computed by DRSQ. The plot is displayed on a Tektronix 4010-1 Graphics Terminal using the routine GRPH2 by G. Shaw as modified by L. Kizer and D. Zambon.

DESCRIPTION:

Location: DP4:BRATCHET

Size:	PLTS.FR	3785 bytes
	PLTS.RB	5302 bytes

PROGRAM USE:

This program is called by PLTO and calls GFDB and GRPH2. It plots spectrum from a disk file computed by DRSQ.

The operator is first prompted for an option to print numerical values of data to be plotted. The data values plotted can be displayed on the operator's terminal, written to the line printer, or not displayed at all. Next, the operator is prompted for the name of the disk file holding the data to be plotted. The file header is read and identifying information from it is displayed. The time slice number corresponding to the desired spectrum is requested and the dB or magnitude option prompted. When the program is signalled by the operator to proceed, it will display the plot. The value of the first component will be the observation energy.

LIST OF VARIABLES:

VARIABLE	TYPE	PURPOSE
OUT	Integer	Switch for data output option.
OUTD	Integer	Channel number.
FILE2	Integer Array	Holds disk file name.
OBSNBR	Integer	Time slice number.
FONNBR	Integer	Number of spectral components.
FRSTDBLK	Integer	First disk block to read.
FRSTDP	Integer	First data point in disk block.
HEADER(57)	Integer	Number of spectral components.
DATA	Real Array	Data to be plotted - passed to GRPH2.
DIST	Real Array	Data read from disk.
IDP	Integer	Switch for dB or magnitude plot.

SWITCH SETTINGS:

SWITCH		SETTING
OUT	=	10 for screen output of plotted data
	=	12 for line printer output
	=	0 for no output of numbers

<u>SWITCH</u>		<u>SETTING</u>
IDP	=	1 for dB versus component number
	=	0 for magnitude versus component number

<u>RELATED PROGRAMS</u>:

PLTA, PLTO, PLTN, PLTT, DRVR, DRSQ, DSTA, DSTN, CHKJ, CHKO, CHOOS, FCTR, GFDB, S128, S64, GRPH2.

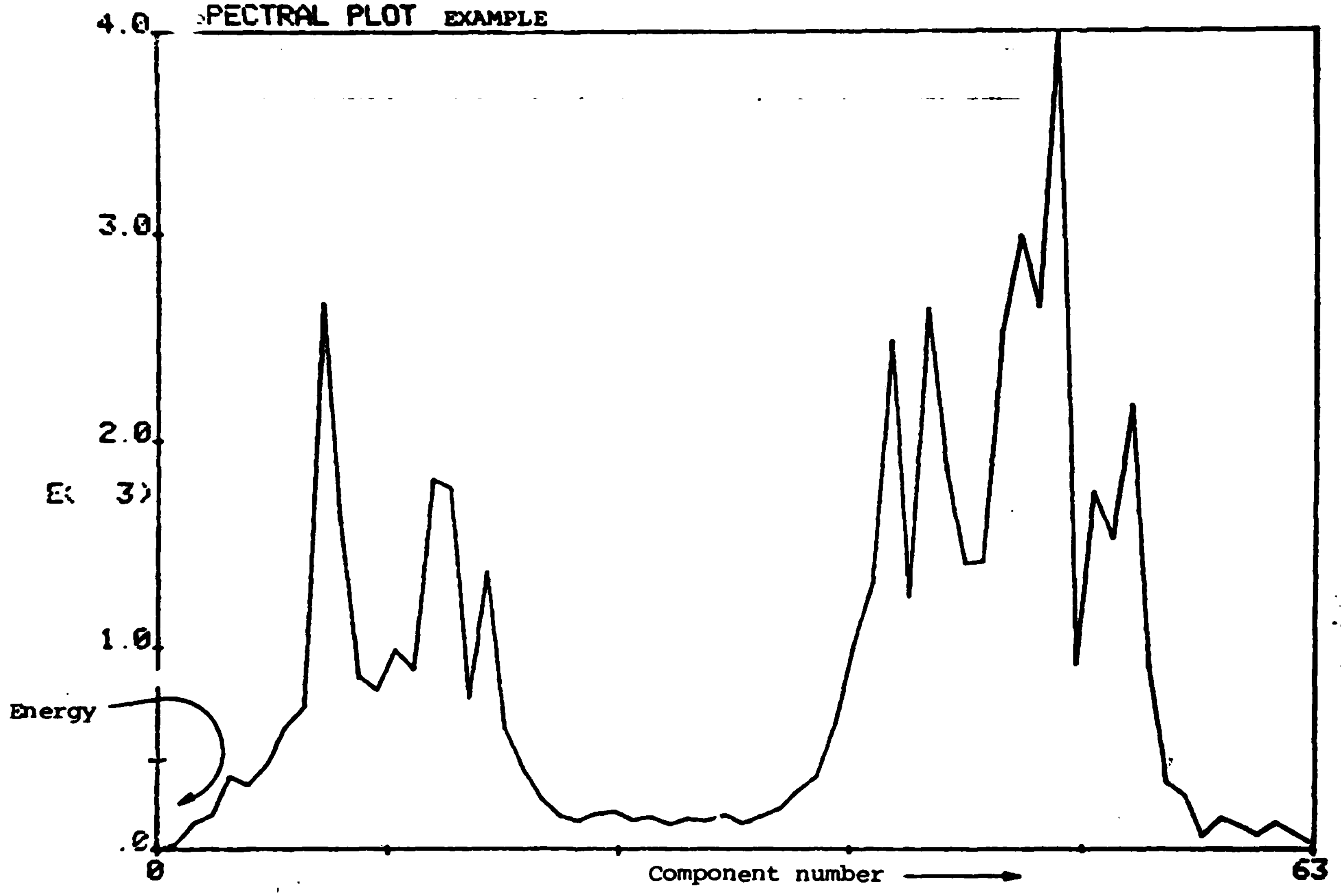


FIGURE 77. Spectral plot of observation number 45. One-hundred twenty-eight (128) point, nonoverlapped Hamming window, energy normalized.

```
C****** ROUTINE PLTS.  THIS ROUTINE IS FOR FORTRAN 5 ! !
C       THIS ROUTINE IS CALLED BY PLTO AND CALLS THE PLOT ROUTINES GRPH2
C       AND GFBD.  GRPH2 WAS WRITTEN BY G.SHAW AND MODIFIED BY L.KIZER
C       AND D.ZAMBON.  IT IS IN THE SPEECH LAB PROGRAM LIBRARY.
C       THIS ROUTINE PLOTS SPECTRUM OF TIME SLICES COMPUTED BY DRS8.
C       THE USER IS PROMPTED FOR THE TIME SLICE NUMBER, FILE NAME,
C       AND OTHER NECESSARY INFORMATION.
C       BY:  CAPT DAN MARTIN
C       DATE:  9/21/82
C       SUBJ:  ACOUSTIC ANALYSIS
C       PLTA ACCOMPLISHES THE FOLLOWING TASKS:
C               1)  GET SPECTRAL FILE
C               2)  DISPLAY INFORMATION FROM HEADER TO IDENTIFY THE FILE
C               3)  GET OBSERVATION NUMBER (TIME SLICE NUMBER)
C               4)  LOCATE DATA IN DISK FILE
C               5)  PREPARE DATA FOR PLOT ROUTINE GRPH2
C               6)  CALL THE PLOT ROUTINE
C       FOR MORE INFORMATION SEE THE USERS MANUAL OR MY THESIS.

        SUBROUTINE PLTS
        OVERLAY OPLTS
        INCLUDE"ARRAYP:F5APS.FR"
        PARAMETER HNDP=512
        PARAMETER QTRBLK=256
        REAL DATA,XMIN,XMAX,DIST
        INTEGER GSPCT,FILE2,HEADER,FSPFN,OUTD,OBSNBR,PHON1,PHONL
        INTEGER FONNBR,FRSTDP,FRSTDBLK,ICNT,D1,IER,OUT

        COMMON / VALS / DATA(HNDP),DIST(512),OBSNBR,FSPFN,OUTD,PHON1
        COMMON / VALS / PHONL,ICNT,IER,FONNBR,FRSTDP,FRSTDBLK,FILE2(13)
        COMMON / VALS / HEADER(QTRBLK)
        COMMON / VALT / D1(37)

        TYPE"****YOU ARE NOW IN ROUTINE PLTS.FR****"
30      TYPE"DO YOU WANT OUTPUT OF DATA PLOTED?"
        TYPE"ENTER:    10 FOR SCREEN OUTPUT OF NUMBERS"
        TYPE"          12 FOR PRINTED NUMERICAL OUTPUT"
        ACCEPT"         0 FOR NO NUMBERS:  ",OUT
        IF(OUT.EQ.0.OR.OUT.EQ.10.OR.OUT.EQ.12)GOTO 40
        TYPE"ERROR:  ILLEGAL INPUT."
        GOTO 30
40      OUTD=10

C****** GET SPECTRAL FILE NAME
50      ACCEPT"**ENTER NAME OF FILE HOLDING SPECTRUM: "
        READ(11,100)FILE2(1)
100     FORMAT(S13)
120     CALL CLOSE(4,IER)
        CALL OPEN(4,FILE2,2,IER)
        CALL CHECK(IER)
        IF(IER.NE.1)TYPE"ERROR ON OPEN OF SPECTRAL FILE, IER= ",IER
        GOTO 180
150     CONTINUE
C****** GET NUMBERS OF 1ST & LAST TIME-SLICES OF SPECTRUM
```

```
180     CALL RDBLK(4,0,HEADER,1,IER)
        CALL CHECK(IER)
        WRITE(OUTD,9110)
        WRITE(OUTD,9100)(HEADER(I5),I5=1,65)
        WRITE(OUTD,9000)(HEADER(I5),I5=14,26)
        WRITE(OUTD,9001)(HEADER(I5),I5=1,13)
        WRITE(OUTD,9004)HEADER(55),HEADER(56)
        ACCEPT"IS THIS THE FILE YOU'RE AFTER? (1=YES/0=NO): ",FSPFN
        IF(FSPFN.EQ.0)GOTO 50
190     ACCEPT"ENTER THE TIME SLICE NUMBER:  ",OBSNBR
        IF(OBSNBR.GE.HEADER(55).AND.OBSNBR.LE.HEADER(56))GOTO 200
        TYPE"ERROR:  THIS NUMBER MUST LIE IN THE RANGE"
        TYPE"[",HEADER(55),",",HEADER(56),"]"
        GOTO 190
200     CONTINUE
240     FONNBR=HEADER(57)
        IF(FONNBR.LE.HNDP)GOTO 260
        TYPE"ERROR:  YOU REQUESTED MORE THAN 512 SPECTRAL COMPONENTS."
        GOTO 200
260     CONTINUE
C****** DISK BLOCK AND ARRAY LOCATION OF 1ST DATA POINT IS
        IA=OBSNBR-HEADER(55)+1
        IB=HEADER(57)*2
        CALL GFDB(IA,IB)
        FRSTDBLK=1+IA
        FRSTDP=(IB-1)/2+1
        CALL RDBLK(4,FRSTDBLK,DIST,4,ICNT,IER)
        IF(IER.NE.9)GOTO 300
        TYPE"READ EOF!  SUCCESSFULLY TRANSFERRED",ICNT,"  QTR BLKS"
        TYPE"PROCEEDING WITH  ",ICNT,"QTR BLOCKS TRANSFERRED."
        IER=1
300     CALL CHECK(IER)
        CALL CLOSE(4,IER)
        CALL CHECK(IER)

        IF(OUT.EQ.12)OUTD=12
C****** MOVE THE DATA UP-FRONT AND INTO THE ARRAY TO PLOT
        ACCEPT"ENTER 1 FOR DB PLOT/ 0 FOR MAGNITUDE:  ",IDP
        IF(IDP.EQ.1)GOTO 310
        DO 305 J1=1,HEADER(57)
305     DATA(J1)=DIST(J1+FRSTDP-1)
        GOTO 320
310     DO 315 J1=1,HEADER(57)
315     DATA(J1)=20*ALOG10(DIST(J1+FRSTDP-1))
320     IF(OUT.EQ.0)GOTO 321

        WRITE(OUTD,9100)(DATA(I5),I5=1,FONNBR)
321     ACCEPT"OK TO CONTINUE? (ENTER ANY NUMBER)  ",I5
322     IF(I5.EQ.0)GOTO 324
324     CONTINUE
        CALL GRPH2("  SPECTRAL PLOT",1,DATA,U,FONNBR,0,XMIN,XMAX,0)
        ACCEPT"OK TO CONTINUE? (ENTER ANY NUMBER)  ",I5
        IF(I5.EQ.0)GOTO 350
350     CONTINUE
```

```
9100    FORMAT(5G12.4)
9110    FORMAT(T5,"HERE'S THE FIRST 65 ELEMENTS OF THE HEADER:")
9000    FORMAT(T3,"SPEECH FILE NAME IS:  ",13S2)
9001    FORMAT(T5,"SPECTRAL FILE NAME WAS:  ",13S2)
9004    FORMAT(T2,"FIRST SPECTRAL TS# =  ",I5,"   AND LAST =  ",I6)

        RETURN
        END
```

FILE:	PLTN
LANGUAGE:	FORTRAN 5
DATE:	September 21, 1982
AUTHOR:	D. Martin
SUBJECT:	Graphics
CALLING SEQUENCE:	PLTN
DATE OF LAST REVISION:	September 21, 1982

PURPOSE:

This routine plots distances from a disk file computed by DSTN. The plot is displayed on a Tektronix 4010-1 Graphics Terminal using the routine GRPH2 by G. Shaw as modified by L. Kizer and D. Zambon.

DESCRIPTION:

Location: DP4:BRATCHET

Size:	PLTN.FR	4132 bytes
	PLTN.RB	5970 bytes

PROGRAM USE:

This program is called by PLTO and calls GFDB and GRPH2. For a specified observation, this routine plots the observation energy as the first data point and the distances at locations corresponding to their phonet numbers. In this way, a sort of scatter plot is obtained.

The operator is first prompted for an option to print numerical values of the data to be plotted. The values plotted can be displayed on the operator's terminal, written to the line printer, or not displayed at all. Next, the operator is prompted for the name of the disk file holding the data to be plotted. The file header is read and identifying information displayed. Then, the observation number is requested. The program will display the plot when signalled by the operator to proceed.

LIST OF VARIABLES:

VARIABLE	TYPE	PURPOSE
OUT	Integer	Switch for data output option.
OUTD	Integer	Channel number.
FIL23	Integer Array	Holds disk file name.
OBSNBR	Integer	Observation number.
FONNBR	Integer	Number of data points (one plus number of phonets).
FRSTDBLK	Integer	First disk block to read.
FRSTDP	Integer	First data point in disk block.
DATA	Real Array	Holds data to be plotted.
DIST	Integer Array	Data read from disk.

SWITCH SETTINGS:

SWITCH		SETTING
OUT	=	10 for screen output of plotted data
	=	12 for line printer output
	=	0 for no output of numbers

RELATED PROGRAMS:

PLTO, PLTS, PLTA, PLTT, DRVR, DRSQ, S64, S128, DSTA, DSTN, CHOOS, CHKJ, CHKO, FCTR, GFDB, GRPH2.

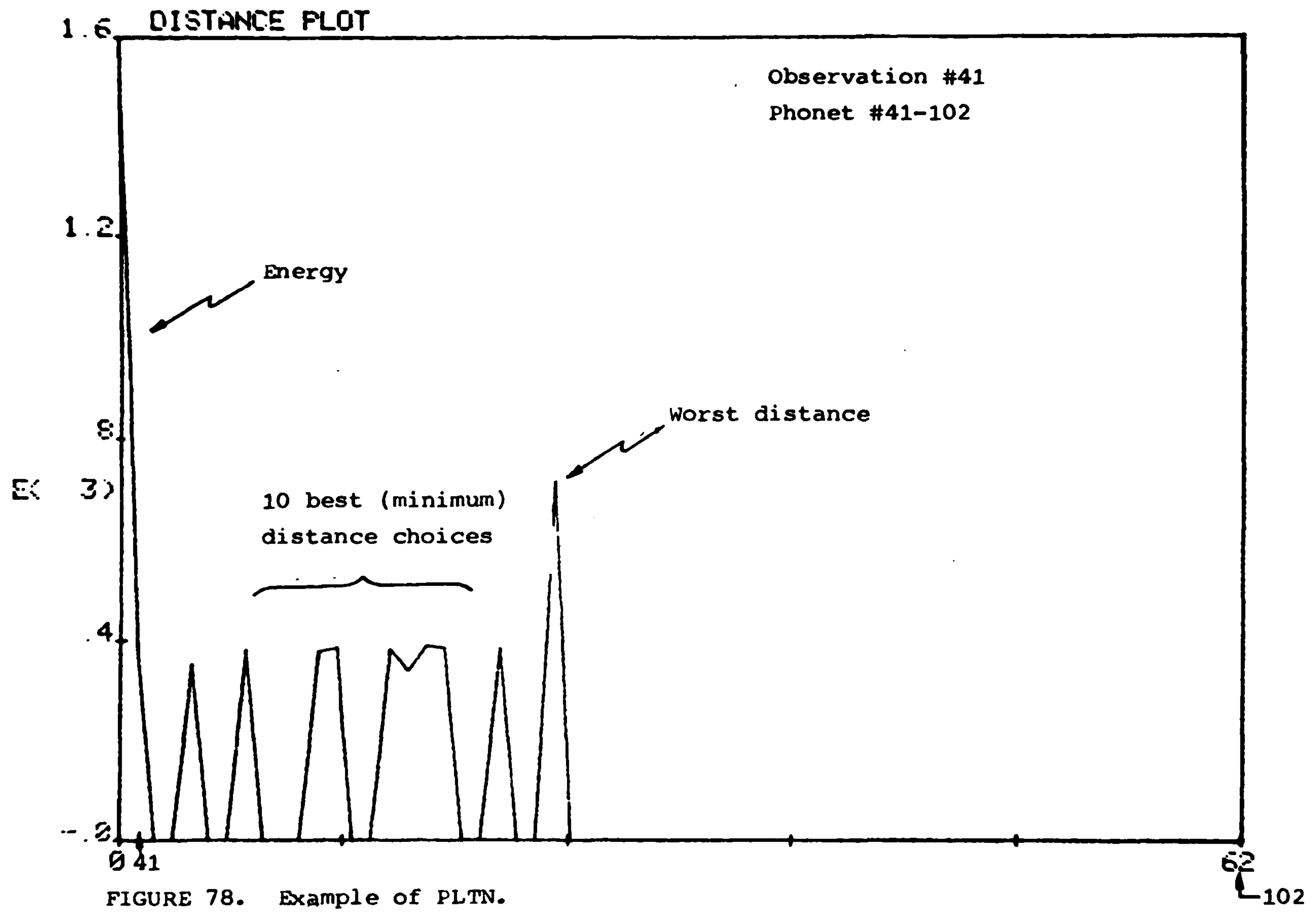


FIGURE 78. Example of PLTN.

```
C****** ROUTINE PLTN.  THIS ROUTINE IS FOR FORTRAN 5 ! !
C       THIS ROUTINE IS CALLED BY PLTO AND CALLS THE PLOT ROUTINES
C       GRPH2 AND GFBD.  GRPH2 WAS WRITTEN BY G.SHAW AND MODIFIED BY
C       L.KIZER AND D.ZAMBON.  IT IS IN THE SPEECH LAB PROGRAM LIBRARY.
C       PLTN PLOTS DISTANCES COMPUTED BY DSTN.  FOR A SPECIFIED
C       OBSERVATION, IT WILL PLOT THE OBSERVATION ENERGY AS THE FIRST
C       DATA POINT, AND THE DISTANCES AT LOCATIONS CORRESPONDING TO
C       THIER PHONET NUMBERS.  IN THIS WAY A SORT OF SCATTER PLOT OF
C       N-BEST DISTANCES IS PLOTTED.
C       BY:  CAPT DAN MARTIN
C       DATE:  9/21/82
C       SUBJ:  ACOIUSTIC ANALYSIS
C       PLTN ACCOMPLISHES THE FOLLOWING TASKS:
C               1)  GET DISTANCE FILE
C               2)  DISPLAY INFORMATION FROM HEADER TO IDENTIFY THE FILE
C               3)  GET OBSERVATION NUMBER AND PHONET NUMBERS
C               4)  LOCATE DATA IN DISK FILE
C               5)  PREPARE DATA FOR PLOT ROUTINE
C               6)  CALL THE PLOT ROUTINE
C       FOR MORE INFORMATION SEE MY THESIS OR THE USERS MANUAL.

        SUBROUTINE PLTN
        OVERLAY OPLTN
        INCLUDE"ARRAYP:F5APS.FR"
        PARAMETER HNDP=512
        PARAMETER QTRBLK=256
        REAL DATA,XMIN,XMAX
        INTEGER GSPCT,FILE3,HEADER,FSPFN,OUTD,OBSNBR,PHON1,PHONL
        INTEGER DIST,FONNBR,FRSTDP,FRSTDBLK,ICNT,D1,IER,D2,OUT

        COMMON / VALS / DATA(HNDP),DIST(1024),OBSNBR,FSPFN,OUTD,PHON1
        COMMON / VALS / PHONL,ICNT,IER,FONNBR,FRSTDP,FRSTDBLK,FILE3(13)
        COMMON / VALS / HEADER(QTRBLK)
        COMMON / VALT / D1(37)
        COMMON / VALU / D2(2054),NCHOICES,NBRPHONES

        TYPE"****YOU ARE NOW IN ROUTINE PLTD.FR****"
30      TYPE"DO YOU WANT OUTPUT OF DATA PLOTTED?"
        TYPE"ENTER:   10 FOR SCREEN OUTPUT OF NUMBERS"
        TYPE"         12 FOR PRINTED NUMERICAL OUTPUT"
        ACCEPT"           0 FOR NO NUMBERS:  ",OUT
        IF(OUT.EQ.0.OR.OUT.EQ.10.OR.OUT.EQ.12)GOTO 40
        TYPE"ERROR:  ILLEGAL INPUT."
        GOTO 30
40      OUTD=10

C****** GET DISTANCE FILE NAME
50      ACCEPT"**ENTER NAME OF FILE HOLDING DISTANCES: "
        READ(11,100)FILE3(1)
100     FORMAT(S13)
120     CALL CLOSE(2,IER)
        CALL OPEN(2,FILE3,2,IER)
        CALL CHECK(IER)
C****** GET NUMBERS OF 1ST & LAST TIME-SLICES OF SPECTRUM
```

```
180     CALL RDBLK(2,0,HEADER,1,IER)
        CALL CHECK(IER)
        WRITE(OUTD,9110)
        WRITE(OUTD,9100)(HEADER(I5),I5=1,65)
        WRITE(OUTD,9000)(HEADER(I5),I5=1,13)
        WRITE(OUTD,9001)(HEADER(I5),I5=14,26)
        WRITE(OUTD,9004)HEADER(40),HEADER(41)
        WRITE(OUTD,9002)(HEADER(I5),I5=27,39)
        WRITE(OUTD,9005)HEADER(42),HEADER(43)
        ACCEPT"IS THIS THE FILE YOU'RE AFTER? (1=YES/0=NO): ",FSPFN
        IF(FSPFN.EQ.0)GOTO 50
190     ACCEPT"ENTER THE OBSERVATION NUMBER:  ",OBSNBR
        IF(OBSNBR.GE.HEADER(40).AND.OBSNBR.LE.HEADER(41))GOTO 200
        TYPE"ERROR:  THIS NUMBER MUST LIE IN THE RANGE"
        TYPE"[",HEADER(40),",",HEADER(41),"]"
        GOTO 190
200     CONTINUE
240     FONNBR=HEADER(43)-HEADER(42)+2
        IF(FONNBR.LE.HNDP)GOTO 260
        TYPE"ERROR:  YOU REQUESTED MORE THAN 512 DISTANCES."
        GOTO 1000
260     CONTINUE
C****** GET DISK BLOCK AND 1ST DATA POINT IS
        IA=OBSNBR-HEADER(40)+1
        IB=4+2*HEADER(47)
        CALL GFDB(IA,IB)
        FRSTDBLK=IA+1
        FRSTDP=IB
        CALL RDBLK(2,FRSTDBLK,DIST,3,ICNT,IER)
        IF(IER.NE.9)GOTO 300
        TYPE"READ EOF!  SUCCESSFULLY TRANSFERRED",ICNT,"  QTR BLKS"
        TYPE"PROCEEDING WITH  ",ICNT,"QTR BLOCKS TRANSFERRED."
        IER=1
300     CALL CHECK(IER)

        IF(OUT.EQ.12)OUTD=12

C****** ZERO-OUT ARRAY DATA
        DO 400 JJ=1,FONNBR+1
400     DATA(JJ)=0.0
C****** FIRST DATA POINT IS OBSERVATION ENERGY
        DATA(1)=DIST(FRSTDP+1)
        TYPE"OBSERVATION NUMBER=",DIST(FRSTDP)
        ACCEPT"IS THIS THE ONE YOU WANT?  (1=YES/0=NO):  ",FSPFN
        IF(FSPFN.EQ.0)GOTO 120
C****** THEN FILL DATA WITH DISTANCES
        DO 410 JJ=FRSTDP+2,3+FRSTDP+2*HEADER(47),2
410     DATA(DIST(JJ)+1)=DIST(JJ+1)
        IF(OUT.NE.0)WRITE(OUTD,9100)(DATA(I5),I5=1,HEADER(43)-HEADER(42)+1)
310     ACCEPT"OK TO CONTINUE? (ENTER ANY NUMBER)  ",I5
        IF(I5.EQ.0)GOTO 312
312     CONTINUE
        CALL GRPH2("  DISTANCE PLOT",1,DATA,U,FONNBR,0,XMIN,XMAX,0)
        ACCEPT"OK TO CONTINUE? (ENTER ANY NUMBER)  ",I5
        IF(I5.EQ.0)GOTO 350
350     CONTINUE
        CALL RESET
9100    FORMAT(5G12.4)
9110    FORMAT(T5,"HERE'S THE FIRST 65 ELEMENTS OF THE HEADER:")
9000    FORMAT(T3,"DISTANCE FILE NAME IS:  ",13S2)
9001    FORMAT(T5,"OBSERVATION FILE NAME WAS:  ",13S2)
9002    FORMAT(T5,"PHONET FILE NAME WAS:  ",13S2)
9004    FORMAT(T2,"FIRST OBSERVATION TS# =  ",I5,"   AND LAST =  ",I6)
9005    FORMAT(T2,"FIRST PHONET TS# =  ",I5,"   AND LAST =  ",I6)

1000    RETURN
        END
```

FILE:	PLTT
LANGUAGE:	FORTRAN 5
DATE:	September 21, 1982
AUTHOR:	D. Martin
SUBJECT:	Graphics
CALLING SEQUENCE:	PLTT
DATE OF LAST REVISION:	September 21, 1982

PURPOSE:

This routine plots an integer disk file in units of from one to 512 integer words on a Tektronix 4010-1 Graphics Terminal using the routine GRPH2 by G. Shaw as modified by L. Kizer and D. Zambon.

DESCRIPTION:

Location: DP4:BRATCHET

Size:	PLTT.FR	2840 bytes
	PLTT.RB	3290 bytes

PROGRAM USE:

This program is called by PLTO and calls GFDB and GRPH2. It will plot segments of an integer disk file; the segment length can be from one to 512 integer words. The routine was designed to be used to plot segments of speech files and to plot distances between several consecutive observations and phonets. The operator is prompted for the disk file name, the segment number, and the segment size. To illustrate, say the header of a file computed by either DRSQ, DSTA, or DSTN is to be examined. Using this routine, the operator could specify the first segment of 256 words along with a print option to examine the header. The plot could be discarded. On the other hand, suppose a curious segment of a speech file is to be examined. One determines which disk block contains the curious segment, and uses PLTT to plot that disk block.

<u>LIST OF VARIABLES</u>:

VARIABLE	TYPE	PURPOSE
OUT	Integer	Switch for data output option.
OUTD	Integer	Unit number.
FILE3	Integer Array	Holds disk file name.
OBSNBR	Integer	Disk file segment.
PHONL	Integer	Disk file segment length.
FRSTDBLK	Integer	First disk block to read.
FRSTDP	Integer	First data point in disk block.
DATA	Real Array	Holds data to be plotted.
DIST	Integer Array	Data read from disk.

<u>SWITCH SETTINGS</u>:

SWITCH		SETTING
OUT	=	10 for screen output of numbers to be plotted
	=	12 for line printer output
	=	0 for no numerical output

<u>RELATED PROGRAMS</u>:

PLTO, PLTA, PLTN, PLTS, DRVR, DRSQ, S128, S64, DSTA, DSTN, CHOOS, CHKJ, CHKO, FCTR, GFDB, GRPH2.

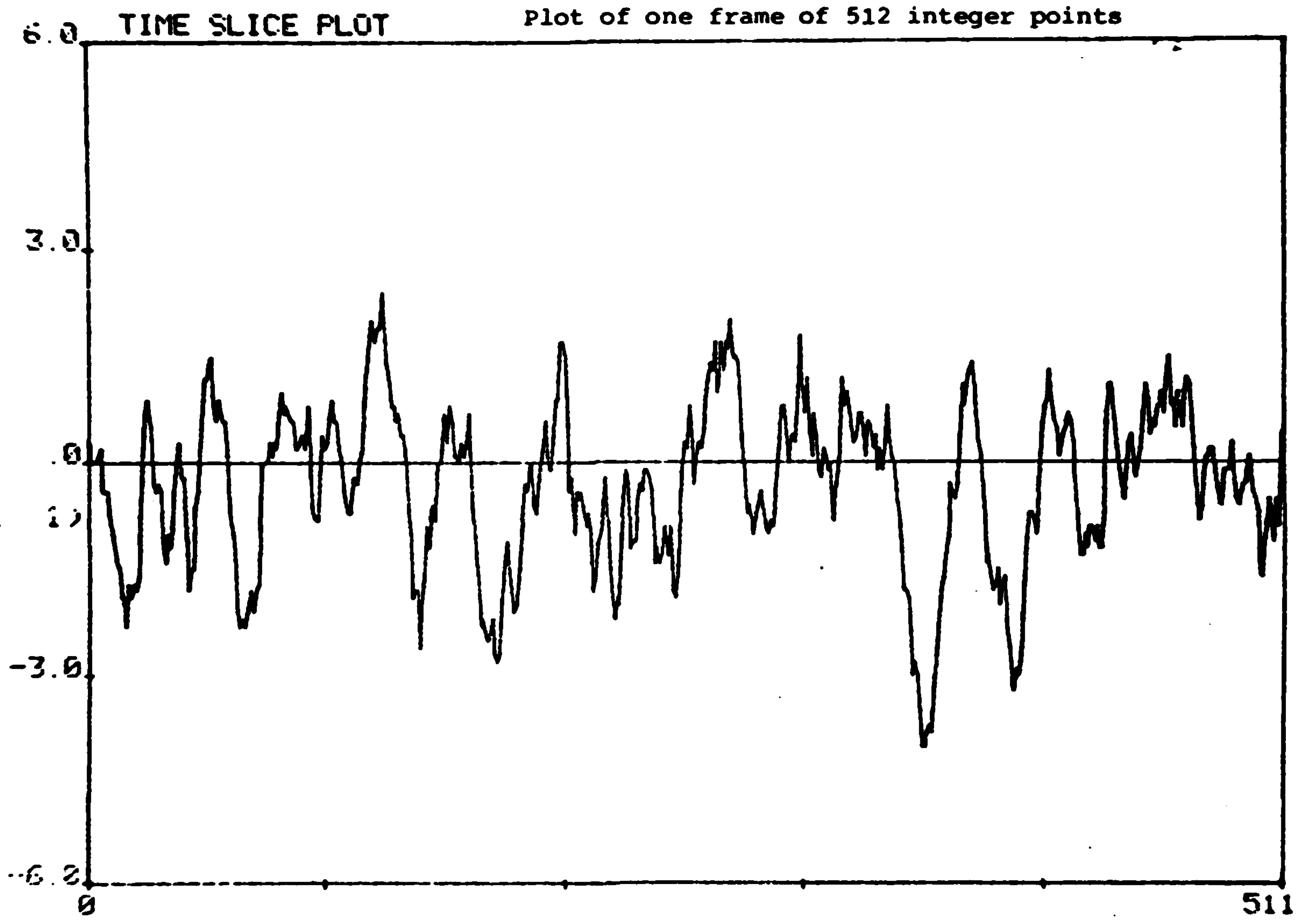


FIGURE 79. Example of PLTT

```
C****** ROUTINE PLTT.  THIS ROUTINE IS FOR FORTRAN 5 ! !
C       THIS ROUTINE IS CALLED BY PLTO AND CALLS THE PLOT ROUTINES GRPH2
C       AND GFBD.  GRPH2 WAS WRITTEN BY G.SHAW AND MODIFIED BY L.KIZER
C       AND D.ZAMBON.  IT IS IN THE SPEECH LAB PROGRAM LIBRARY.
C       THIS ROUTINE PLOTS INTEGER DISK FILES.  IT WILL PLOT SEGMENTS
C       OF THE FILE; EACH SEGMENT IS SPECIFIED BY ITS NUMBER AND POINT
C       LENGTH.  THE POINT LENGTH CAN BE FROM 1 TO 512 INTEGER WORDS.
C       THE USER IS PROMPTED FOR THIS INFORMATION AS WELL AS FOR THE
C       FILE NAME.
C       BY:  CAPT DAN MARTIN
C       DATE:  9/21/82
C       SUBJ:  ACOUSTIC ANALYSIS
C       PLTA ACCOMPLISHES THE FOLLOWING TASKS:
C               1)  GET FILE
C               3)  GET UNIT NUMBER AND UNIT SIZE
C               4)  LOCATE DATA IN DISK FILE
C               5)  PREPARE DATA FOR PLOT ROUTINE GRPH2
C               6)  CALL THE PLOT ROUTINE
C       FOR MORE INFORMATION SEE THE USERS MANUAL OR MY THESIS.

        SUBROUTINE PLTT
        OVERLAY OPLTT
        INCLUDE"ARRAYP:F5APS.FR"
        PARAMETER HNDP=512
        PARAMETER QTRBLK=256
        REAL DATA,XMIN,XMAX
        INTEGER GSPCT,FILE3,HEADER,FSPFN,OUTD,OBSNBR,PHON1,PHONL
        INTEGER DIST,FONNBR,FRSTDP,FRSTDBLK,ICNT,D1,IER,OUT

        COMMON / VALS / DATA(HNDP),DIST(1024),OBSNBR,FSPFN,OUTD,PHON1
        COMMON / VALS / PHONL,ICNT,IER,FONNBR,FRSTDP,FRSTDBLK,FILE3(13)
        COMMON / VALS / HEADER(QTRBLK)
        COMMON / VALT / D1(37)

        TYPE"****YOU ARE NOW IN ROUTINE PLTT.FR****"
30      TYPE"DO YOU WANT OUTPUT OF DATA PLOTTED?"
        TYPE"ENTER:   10 FOR SCREEN OUPUT OF NUMBERS"
        TYPE"         12 FOR PRINTED NUMERICAL OUTPUT"
        ACCEPT"          0 FOR NO NUMERICAL OUTPUT:  ",OUT
        IF(OUT.EQ.0.OR.OUT.EQ.10.OR.OUT.EQ.12)GOTO 40
        TYPE"ERROR:  ILLEGAL INPUT."
        GOTO 30
40      OUTD=10

60      CONTINUE
C****** GET INTEGER FILE NAME
        ACCEPT"**ENTER NAME OF FILE HOLDING INTEGER DATA: "
        READ(11,100)FILE3(1)
100     FORMAT(S13)
120     CALL OPEN(5,FILE3,1,IER)
        CALL CHECK(IER)
190     ACCEPT"ENTER THE TIME SLICE NUMBER:  ",OBSNBR
220     ACCEPT"ENTER THE NUMBER OF POINTS IN TIME SLICE:  ",PHONL
        IF(PHONL.LE.HNDP)GOTO 260
        TYPE"ERROR:  YOU REQUESTED MORE THAN 512 POINTS."
        GOTO 220
260     CONTINUE
C****** DISK BLOCK THAT HOLDS 1ST DATA POINT IS
        IA=OBSNBR
        IB=PHONL
        CALL GFDB(IA,IB)
        FRSTDBLK=IA
        FRSTDP=IB
        CALL RDBLK(5,FRSTDBLK,DIST,4,ICNT,IER)
        IF(IER.NE.9)GOTO 300
        TYPE"READ EOF!  SUCCESSFULLY TRANSFERRED",ICNT,"  QTR BLKS"
        TYPE"PROCEEDING WITH  ",ICNT,"QTR BLOCKS TRANSFERRED."
        IER=1
300     CALL CHECK(IER)
        CALL CLOSE(5,IER)
        CALL CHECK(IER)

        IF(OUT.EQ.0)GOTO 310
        IF(OUT.EQ.12)OUTD=12
C****** MOVE THE DATA UP-FRONT AND INTO THE ARRAY TO PLOT
310     IU=PHONL
        DO 320 J1=1,IU
320     DATA(J1)=DIST(J1+FRSTDP-1)
        IF(OUT.EQ.0)GOTO 322
        TYPE"OUTD=",OUTD
        WRITE(OUTD,9100)(DATA(I5),I5=1,IU)
322     ACCEPT"OK TO CONTINUE? (ENTER ANY NUMBER)  ",I5
        IF(I5.NE.0)GOTO 324
324     CONTINUE
        CALL GRPH2("  TIME SLICE PLOT",1,DATA,U,IU,0,XMIN,XMAX,0)
        ACCEPT"OK TO CONTINUE? (ENTER ANY NUMBER)  ",I5
        IF(I5.EQ.0)GOTO 350
350     CONTINUE
9100    FORMAT(5G12.4)

        RETURN
        END
```

FILE:	CHKO
LANGUAGE:	FORTRAN 5
DATE:	September 21, 1982
AUTHOR:	D. Martin
SUBJECT:	Acoustic Analysis
CALLING SEQUENCE:	CHKO (IER)
DATE OF LAST REVISION:	September 21, 1982

PURPOSE:

This routine is called by DSTA and DSTN, and calls no subroutine. It checks the value of IER returned from an OPEN file attempt. It returns a caution statement if the unit number is in use.

DESCRIPTION:

Location: DP4:BRATCHET

Size:	CHKO.FR	586 bytes
	CHKO.RB	288 bytes

LIST OF VARIABLES: None.

ARGUMENT STRUCTURE:

ARGUMENT	TYPE	PURPOSE
IER	Integer	Holds error code from OPEN system call.

PROGRAM USE:

To be called by DSTA and DSTN to provide caution statement.

RELATED PROGRAMS:

PLTO, PLTA, PLTS, PLTT, PLTN, FCTR, GRPH2, GRDB, DRVR, DRSQ, DSTA, DSTN, CHKJ, CHOOS, S128, S64.

```
C****** ROUTINE CHKO.  THIS ROUTINE IS FOR FORTRAN 5 ! !
C       THIS ROUTINE IS CALLED BY DSTA AND DSTN, AND CALLS NO SUBROUTINES.
C       IT CHECKS THE VALUE OF IER RETURNED FROM AN OPEN FILE ATTEMPT.
C       IT RETURNS A CAUTION STATEMENT TO THE SCREEN IF THE UNIT NUMBER
C       IS IN USE.
C       BY:  CAPT DAN MARTIN
C       DATE:  9/21/82
C       SUBJ:  ACOUSTIC ANALYSIS
C       FOR MORE INFORMATION SEE THE USERS MANUAL FOR DSTA, DSTN, OR THIS
C       ROUTINE.  OR SEE MY THESIS.

        SUBROUTINE CHKO(IER)
        OVERLAY OCHKO
        INTEGER IER
        IF(IER.NE.3096)GOTO 100
        IER=1
        TYPE"******CAUTION:  UNIT NUMBER IS IN USE!******"
100     RETURN
        END
```

VITA

Dan Martin was born on 24 March 1952 in Wheeling, West Virginia. He graduated from Walnut Ridge High School in Columbus, Ohio, in 1970 and enlisted in the U.S. Air Force that same year. While assigned to the Air Force Flight Dynamics Laboratory, Wright-Patterson AFB, Ohio, he attended Wright State University in Fairborn, Ohio. He majored in Applied Mathematics and upon graduation in December 1976, he received his B.S. degree. He was commissioned upon his graduation from Officers Training School at Lackland AFB, Texas, on 24 March 1977. He attended Communications-Electronics Officers Course, Keesler AFB, Mississippi from April 1977 to December 1977. He was then assigned to the USAF Frequency Management Office, Bolling AFB, Washington, D.C., and served as Radio Frequency Engineer. In August 1980, he was assigned to the School of Engineering, Air Force Institute of Technology, to pursue a B.S.E.E. degree. Upon his graduation in March 1982, Captain Martin was selected to remain at the Air Force Institute of Technology to pursue an M.S.E.E. with emphasis in Communications and Electromagnetic Field Theory. He is a member of Tau Beta Pi and Eta Kappa Nu, and the father of one son in whom he takes great pride. The author's permanent address is:

1236 Huntly Drive
Columbus, Ohio 43227